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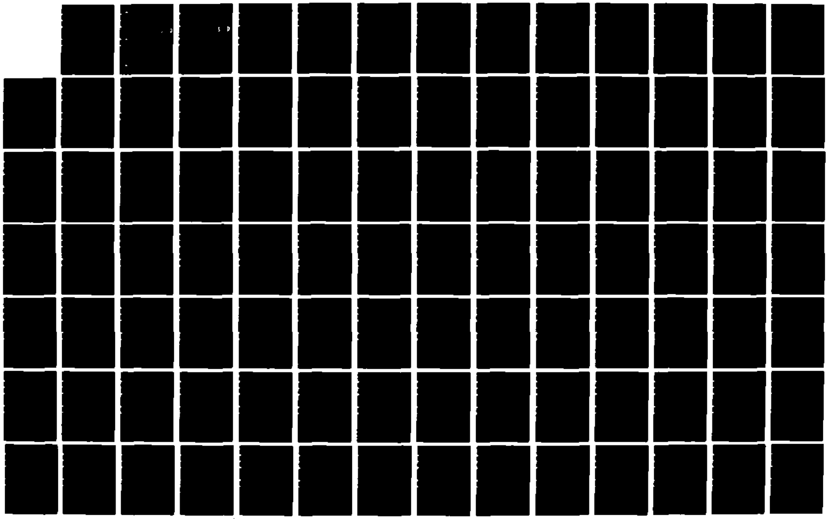

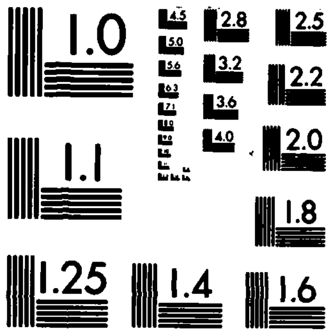

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ANALYSIS OF INTEGRATED AND NONINTEGRATED VOICE

AND DATA NETWORKS FOR DOD COMMUNICATIONS

THESIS

Stephen L. Walker
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AFIT/GCS/ENG/85S-2

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ANALYSIS OF INTEGRATED AND NONINTEGRATED VOICE AND DATA NETWORKS FOR DOD COMMUNICATIONS

THESIS

Presented to the Faculty of the School of Engineering

of the Air Force Institute of Technology

Air University

In Partial Fulfillment of the

Requirements for the Degree of

Master of Science in Electrical Engineering

Stephen L. Walker, B.S.

Captain, USA

September 1985

Preface

The purpose of this study was to perform a network analysis for determining the best means for DOD communications. Due to difficulty in determining DOD communications requirements it was necessary to modify the scope of this research to the analyzation of specific parameters and data from the Defense Data Network. The immediate need presented by this research was the importance of DOD research to instigate intensive studies into the area of voice and data integration as a viable approach to future communications.

I am deeply indebted to my faculty advisor, Major Walter Seward, for his continuing patience and instruction. I also want to thank Capt David A. King, my reader. I special word of thanks to Capt Mike Adams for assisting me in a follow-on effort of his thesis. A word of thanks to Mr. John Salerno for information and cooperation in collection of performance criteria and the analysis of the Simplified Voice Trunking Model. I especially wish to thank my wife, Lula, and my son, David Lee, for their understanding and concern during the long hours of study. Finally, I thank my Lord and Savior with whom all things are possible.

Stephen L. Walker

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Abstract

This analysis determined the best approach and switching technique for use in DOD communications. The approach and switching techniques considered were integrated and nonintegrated voice and data networks using circuit, packet, and hybrid switching. The Simplified Voice Trunking Model(SVTM) from Rome Air Development Center was used as the network using hybrid switching for voice and data integration.

The analysis was accomplished with a network analysis of various performance criteria using different approaches to voice and data communication networks. The network analysis was performed using mathematical analysis and simulation models. The simulation models created were a circuit switched model for analog voice, a packet switched model for digital data, a packet switched model with multiple arrivals for digital voice, interactive data and bulk data, and a packet switched model for digital voice and interactive data. The simulation models allowed for changes in percentage of voice and data, arrival rates, service rates, path algorithm, percentage of interactive versus bulk data, queue lengths and distance of message travel. The results of the network analysis were also compared to the results of the Simplified Voice Trunking Model. The results of this analysis indicate that DOD needs to instigate intensive research into integrated voice and data communications using packet and hybrid switching techniques. This analysis showed the best approach to DOD communications was with use of voice and data integration and hybrid switching as used in the Simplified Voice Trunking Model.

ANALYSIS OF INTEGRATED AND NONINTEGRATED VOICE AND DATA NETWORKS FOR DOD COMMUNICATIONS

I. Introduction

Background

Voice and Data Integration. The integration of voice and data over the same communications network can make future communications more efficient. The demand for the integrating of voice and data over a common communications link is brought about by advances in technology and the need for more communications capability. A specific technological advance which sparked interest in voice and data integration was the digitization of voice. Two additional factors which have increased interest in voice and data integration are the poor utilization of separate voice and data networks and the success and increasing need of data communications(39:1).

Recent advances in speech encoding provide a link between the methods used in data communications and the need to improve voice network performance. A normal voice conversation is separated into periods of speaking (talkspurts) and periods of silence, it is a good candidate for packetization(39:1,3). In an experiment by C. Weinstein and J. Forgie (63), voice signals were successfully digitized and transmitted over the ARPANET using packet switching. Voice digitization has advantages such as better encryption and decryption capabilities, less noise interference and increased

circuit utilization(30:1.3).

Some present communications networks have low utilization. An example of a poorly utilized communications network is the present day telephone system. The telephone system is a circuit switched network which uses analog signals for the transmission of a voice conversation. Studies completed as early as 1959, by Bell System, show the average voice conversation uses only 35 to 40 percent of the circuit(12). When a circuit switched network is used for analog voice, network resources are wasted.

Increased utilization and improved use of resources can be seen in digital data communications. Data communications are digitized for transmission over a communications path. This data can be sent as a complete message or broken down into smaller portions, called packets. The message travels through a system from node to node queueing at each node until the next path connection is complete(64:116). This technique is called message or packet switching. Packet switching allows communications paths in a system, not immediately in use by one user, to be available for other traffic.

Communications services worldwide are analyzing the use of a voice and data integrated networks. The Department of Defense(DOD), however, has directed efforts towards separate voice and data networks for the Defense Communications System(DCS). The DCS will use the Digital Switched Network(DSN) for voice and the Defense Data Network(DDN) for data(2:2). The present Worldwide Digital System Architecture(WWDSA) which directs future planning also calls for separate voice and data networks(50). The present direction of the

DOD is not the same as that of the rest of the world. Although the DCS serve a different user population and have different goals than that of its commercial counterparts, it is impractical to ignore the possiblility of integrating voice and data when the rest of the world sees the advantages of such a system. Mr. M.J. Ross expresses this idea with a different outlook in his article on Military/Government communication:

> "There is a tendency to continually upgrade communications to the highest technology possible, in order to derive the maximum advantage in capability over a potential adversary."(56:19).

The factors of superiority and survivability make it necessary to research all possible solutions to DOD communications to determine if it is necessary to integrate voice and data or not. A study of this nature was completed in 1973 by Gitman and Frank(31). Gitman and Frank evaluated switching strageties for integrated DOD voice and data networks and determined that an integrated voice and data packet switched network was the best possible approach to DOD comunications.

Inherent with new communcations systems, such as voice and data integration, are new communications problems. These problems exist in the area of network control and selection of switching strategies. Network control is keyed by the flow of traffic entering the network. The individual characteristics of voice and data make different demands on a communication system. Flow control, for instance, addresses the different tolerances of voice and data for the effects of delay and error. Voice is more sensitive to the effects of delay, whereas data is hindered more by error(38:1006). The selection of the appropriate switching technique depends on the goals of the

network. Rapid advances in technology and increased DOD communication needs have motivated research to upgrade present communications. The majority of research for DOD communication has been directed towards separate packet switched networks. The switching techniques must meet the requirements of the user and the characteristics of voice and data.

Switching Strategies. The different approaches to voice and data communications depend greatly on the method of switching scheme used. Circuit, packet and hybrid switching techniques offer different means of transfering information through a communications network. Circuit switching establishes a complete end to end communications path prior to the transmission of information over a circuit. Packet switching breaks a message into smaller packets and moves them node by node through the network. Hybrid switching is a combination of a circuit and packet switching. The complexity of DOD communications demands selection of the switching technique which best fits its specific requirements. This requires analysis of all possible switching strategies, including the complex and flexible hybrid technique. This research provides an analysis of possible approaches available for future DOD communications.

Statement of Problem and Scope

The primary question to be addressed in this ~~research~~ thesis is, "What is the best switching approach, with or without voice and data integration, that will maximize mission essential needs of DOD communications?" This question can be answered by modeling

integrated voice and data networks and separate voice and data networks. Included in this modeling will be the analysis of different switching techniques to determine the best switching approach. Mathematical analysis results are used to verify a simulation model. This simulation model is used to analyze different possibilities of meeting DOD requirements. These DOD requirements and performance criteria for switching are defined and presented in Chapter II. Specific conditions are set to insure performance changes are based on the switching techniques, and the percentages of voice and data in a system.

Approach

The approach taken in this research was to use analytical techniques and to develop a simulation model for analyzing different voice and data integrated and nonintegrated environments. The first step in this research was to create a mathematical model to evaluate integrated and nonintegrated voice and data networks. This included selecting appropriate performance measures and evaluation techniques to analyze system requirements. Once a mathematical model was created an evaluation of realistic system parameters to determine performance was calculated. The next step was to evaluate simulation techniques to determine an appropriate simulation language to use in modeling selected networks. Simulation models were created based on performance criteria used in the mathematical analysis. The simulation model was then compared to the mathematical model and actual system for verification and validation of simulation results.

The simulation model was used to make a performance evaluation of network parameters which are too difficult to analyze mathematically. After a complete analysis of all DOD requirements, a network model which best meets system requirements was selected and recommended as the direction for future communications.

Sequence of Presentation. Chapter II details network performance evaluation and measurement techniques. It includes performance measures, evaluation methods, and a mathematical analysis of three network designs using example data. Chapter III discusses the procedures involved in making a simulation model. This chapter includes the selection of a simulation language and the formulation of simulation models based on the analysis of chapter II. Chapter III also discusses verification and validation of the simulation models. Chapter IV is the comparison of the three network topologies using simulation models and various performance parameters. Chapter IV includes a comparison of the network topologies to the simulation study, Simplified Voice Trunking Model(SVTM), of the Rome Air Development Center(20). Chapter V includes research results, specific conclusions, and recommendations for future investigation.

Literature Research Results

Extensive research has been done in the area of voice and data communications. The majority of research, whether for integrated or nonintegrated voice and data, has been done with reference to the three switching techniques circuit, packet and hybrid.

Circuit Switching. The most commonly used means of voice communication today, the telephone system, is an example of circuit switching. Circuit switching establishes a complete end-to-end communication path prior to the transmission of information over the circuit. Two major problems which occur with circuit switching are wasted time to secure a communications path and wasted time during transmission. Since circuit switching requires a complete path before transmission, significant time is wasted while securing the entire path. Transmission capacity is wasted because circuit switching of voice uses approximately 40 percent of the entire channel capacity(10:1479). The greatest advantage of circuit switching over other switching techniques is the more efficient handling of extremely long messages.

Packet Switching. "Contrary to circuit switching, packet switching involves moving information from place to place on an as-needed basis, where the amount of information and the end points change with time."(56:3-4). Packet switching is a very popular alternative to circuit switching for data communications. This form of data communications has grown rapidly since the experimental packet switched ARPANET was established in 1966 to connect the System Development Corporation and MIT Lincoln Laboratory(2:5). The experimental packet switched ARPANET now services institutions all over the United States (2:5). The packet switching form addressed in this thesis is also called "store and forward" switching. Voice and data are digitized, broken down into packets and routed through the network using store and forward procedures(39:8). A packet is sent

from node to node and stored in a queue at each intermediate node, then forwarded to the next node until reaching its final destination. Digitized voice and data packets are queued at intermediate nodes until an outgoing channel is available. Packet switching provides higher channel utilization and greater flexibility than circuit switching. Packet switching has two significant limitations affecting its usage. The first of these limitations, lessened by the rapid advances of technology, is the need for substantial computer processing (36:25). More complex computer processing is required in packet switching than circuit switching because of the need to control intermediate switching nodes. The other limitation is real time traffic delay characteristic of digitized voice conversations. Delay of voice packets must be very low in order to maintain the quality of speech acceptable to meet user requirements. Although a tradeoff exists when considering voice packetization, speech quality is the ultimate goal. Minimizing delay requires decreasing the packet size, however high utilization requires increasing the packet size(63:697). In the ARPANET experiment by C. Weinstein and J. Forgie(68), the largest packet size which the network could consistently transmit was 100ms-200ms in length with five to ten packet arrivals per second(68:967). This study proved the possibility of transmitting digitized voice over a packet switching communications link. With rapidlly improving techniques there is a high probability of efficiently digitizing voice over a packet switched network. Packet switching has the potential of providing efficient communications for voice and data(31:1568). Packet switching offers significant advantages with lower delay times,

reduced network costs, and the ability to adjust to various message sizes. The key disadvantage of packet switching is the transmission of extremely long messages. As the message size increases so does the buffer capacity and the probability of packet loss.

Hybrid Switching. Hybrid switching, the combination of circuit and packet switching, can also be a good alternative. The use of hybrid switching allows a circuit switched path for voice and a packet switched path for data. The benefit of this technique is the use of the circuit switched path for the transmission of data packets when voice is not being transmitted. There are several methods of integrating circuit and packet switching. The method discussed in this study is the multiplexer approach discussed by Gitman(29). The multiplexer approach uses a single channel between all nodes. This channel is partitioned into frames using synchronization. Each frame is then divided into two separate regions to accommodate circuit switched traffic in one and packet switched traffic in the other(29:1290). The multiplexer approach integrates digitized voice and data using circuit switching in one frame and digitized data by packet switching in the other frame. This configuration allows data traffic to be transmitted over the unused portion of channel capacity normally dedicated to voice traffic(71:696-697). The use of the idle time found in the circuit switched frame leads to a reduction in packet delay time. This approach to hybrid switching utilizes the best advantages of circuit and packet switching and acquires some new disadvantages. If data is not sent over the circuit switched path then the network maintains the circuit switched characteristic of

wasted resources and higher delay time. On the other hand, if data is integrated on the voice path the network requires a more complicated method of controlling the data flowing over the communication path.

Conclusion. There has been a great deal of research concerning voice and data integration and the various switching techniques. An analysis of the voice and data integrated networks versus separate voice and data networks using the different switching techniques was necessary to determine the best approach to voice and data communications networks. In order to accomplish this task it was important to establish the performance criteria to be evaluated and the method of evaluation. Chapter II discusses the necessary criteria and evaluation techniques for determining the best means of voice and data communications.

II. Performance Analysis

This chapter discusses performance evaluation and measurement involved with making integrated and nonintegrated voice and data network models. The procedures accomplished in the chapter are two fold. The first part discusses performance criteria and measurement techniques necessary to analyze delay, throughput, and cost in voice and data networks. The second part of this chapter uses specific network data and measurement techniques to analyze the performance of three specific network topologies. This mathematical performance will be used to verify the accuracy of simulation models in a later chapter.

Performance Evaluation Criteria

Network performance is usually measured in terms of throughput, delay and cost. Voice and data integrated networks are specifically analyzed, in the research, using quality of speech and loss of speech packets. Other performance checks considered in this evaluation are buffer capacity and utilization. The ultimate goal in network performance is to achieve high throughput, low delay and low cost. Based on present day technology the ultimate goal is rather unrealistic to achieve, because lower delay and higher throughput lead to higher cost. The relationships of these three network performance measures will be analyzed with respect to integrated and nonintegrated voice and data networks.

Throughput is based on channel capacity, packet overhead and service rate. Service rate is directly proportional to channel capacity and overhead. Total delay is broken down into queuing delay, packet delay, and circuit delay. Queuing delay is the time a packet waits to be processed at a particular node. Packet delay is the time to service a packet, and transmit it to the next node. Circuit delay time is the time it takes to set up the entire message path, the time to signal the message transmission to begin, and the time to transmit the message and release the transmission path. Cost analysis can be approached in many ways. In this analysis cost performance included switching costs, tariff (mileage) costs and voice digitization costs. Switching cost was a set cost based on the number and type of each switch in the network. Tariff (mileage) cost was a base rate cost plus incremental node to node milage costs. Voice digitization cost is based on the rate of digitization. It is important to remember that although cost is important, meeting the requirement of the network is the key concern.

There are two key factors which affect speech encoding in communications networks. These two factors are speech quality and speech packet loss(33). The quality of speech is primarily determined by the voice digitization rate used to send voice packets over a communication path(13:109). Speech quality is also affected by the prioritization used for a particular circuit. The second factor, the loss of speech packets can be detrimental to a conversation. The loss of speech packets depended on buffer capacity, prioritiztion of voice and data , and the service rate. In order to convert from the analog voice communications to digitized

voice communications speech quality must be as high or higher than presently used analog voice networks.

The other two performance criteria which affect network performance are buffer capacity and utilization. Buffer capacity affects the packet loss of a network. In this study buffer capacity was kept as small as possible without increasing speech packet loss to unacceptable levels. Utilization is another tool used to check the network. Utilization is a ratio of arrival rates to service rates at a node and shows whether a communications path is efficently used or not.

Three important categories of performance criteria are user demands, manager demands and designer demands (40:18). Although the demands of these three categories are different their ultimate goal is similar. The users view emphasizes friendliness, speed, low cost, and security. The manager view is that of a cost-performance tradeoff oriented towards high utilization, throughput, and power. Of the three demands the designers view gives the more unbiased point of view. The designer is concerned with efficiency and accuracy. With the ability to vary certain input parameters, within the simulation model, it is possible to determine the effect that specific performance criteria have on each of the three views. Maximization of each view will show the effects that changes in one view have on the other views.

User View. The user's view of a network is quite different from that of the manager or designer. In order to meet user needs it is important to first reduce total delay time. The reduction of delay

can occur in several ways. Queuing delay can be reduced by increased communications paths to prevent bottlenecks, however this increases system cost. This type of delay is also decreased by faster service rates or slower arrival rates. Packet delay time is comprised of two factors, packet service time and the time required for a packet to travel from node to node. Packet service time is the time required at a node to process a packet. The time to service a packet is based on the node service rate and the packet size. The other factor of packet delay time is affected by the distance a packet must travel. Depending on circuit use a packet may have to take a longer route to its destination. In this study, an assumption is made that a certain percentage of packets take the shortest path to destination and the rest of the packets take the shortest path plus an additional hop to destination. An increased percentage of the shortest path versus one or more additional hops reduces packet delay. Packet delay, as well as circuit delay is reduced by decreasing the header, packet, or message size. Reducing the header reduces the packet size without increasing the number of packets and therefore decreases delay. A decrease in packet size without decreasing message size leads to an increased number of packets. This can increase the probability of packet loss in a situation with small queue lengths.

Another important consideration for the user is the ease with which a network can be used. Two key parameters which affect this concept are header size and availability. Increased header size yields a friendlier network. ""Friendliness" means how easy a network is to interact with(40:19)". When header size is increased so is packet size and packet delay. Availability on the other hand

can lead to more communication paths or faster service rates, therefore increased costs.

Security and voice digitization rates are also key considerations of the user. Security denies unauthorized access to a users data and transmissions. It is easier to secure a digitized voice than it is to secure analog voice. "Analog speech can be scrambled, but the sophistication and low cost of digital encoding techniques make the digital more attractive(13:113)." Security is also affected by the rate at which voice is digitized. The lower the voice digitization rate the easier it is encrypt and decrypt a channel. Encryption/decryption is easier because there is less to encrypt or decrypt(43:292). Low voice rates also provide greater storage capability. At 64Kbps a device with 640K bits of memory can store ten seconds of digitized speech, where at 800bps the same device can store 13 minutes of digitized speech(13:113). Problems arise as voice digitization rates decrease. As voice rates decrease so does speech quality. In voice networks speech quality is an important consideration for the user(40:19). There are many user tradeoffs and considerations when designing a voice and data network. Many contradictions are seen while analyzing the user's point of view. The user's point of view is summarized in Table 1.

TABLE 1

USER DEMAND TRADEOFFS IN A VOICE AND DATA INTEGRATED NETWORK

Demand	Tradeoff
decrease delay	increase cost and/or communication paths increase service rate
increase header	increase friendliness, increase cost, increase delay, increase packet size
decrease queue size	increase probability of packet loss decrease cost of nodes
decrease packet information size	increase queue size decrease packet service time
increase availability	increase service rate and/or communication paths, increase cost
decrease voice digitization rate	increase security(encrypt/decrypt), decrease speed quality, increase probability to send voice and data over same the channel circuit, increase number of voice channels over the same circuit.

Manager View. From a network manager's point of view, an important goal is the maximization of the use of network resources(40:19). Two key performance measures of importance to the manager are throughput and delay. A ratio of throughput to total delay gives the best management tool for performance measurement, power (40:19). The manager seeks to achieve maximum power by increasing throughput or reducing delay. Power is also beneficial to the needs of the user because increased power requires reduced

delay for given throughput.

Voice packet loss is another measure used by the manager. Due to the characteristics of voice communication, excessive packet loss cannot be tolerated. When too many voice packets are lost the quality of speech degrades to an unintelligible level. Voice packet loss can occur when a packet attempts to enter a full queue and gets kicked out of the system. The problem of packet loss can be solved in three ways. The first way to stop packet loss is to use larger queues, however this is quite expensive. The second means of preventing speech packet loss is with the use of a priority system of speech over data. Problems still occur if the queue is filled with voice packets and another voice packet attempts to enter the queue. This method not only causes the loss of a voice packet but possibly a larger number of data packets. The final means of solving the speech packet problem is to use a dedicated path for voice packets.

Three approaches are discussed in this study which permits analysis of different criteria critical to network design. Even though the manager is concerned about the user he is more dedicated to network performance. An example of this can be taken from the users view discussion of increased overhead. There is a tradeoff between the user and manager concerning overhead and delay. The user desires the network to be as friendly as possible without bringing delay too high. The manager though concerned about user friendliness wants to keep delay to a minimum. Both user and manager seek to improve the network, even though these approaches differ.

<u>Designer View.</u> The designer compares the actual network

performance with the predicted performance(40:20). This includes emphasis on efficiency and effectiveness of performance criteria. An analysis of network efficiency shows areas which can be changed to improve overall performance. The efficiency of specific parts of the network are analyzed to determine performance criteria minimums and maximums. One particular method to check efficiency is by running the simulation model to find the minimum queue capacity necessary for desired output at each node. This type of evaluation is important because of the large cost involved with increased buffer capacity. A mathematical simulation analysis helps enhance network performance and keep cost to a minimum. The designer uses simulation models to check efficiency and accuracy of networks which are too difficult to check mathematically. A mathematical analysis can be used to check specific performance areas and to verify the output of a simulation model. In an integrated voice and data network the designer, like the manager, is concerned with speech packet loss. The designer must use speech packet loss probability when checking the efficiency of the network. Both time interval between packets and the size of packets are important factors in reducing speech packet loss(68:963). These factors can be adjusted easily with a simulation model by changing other criteria.

Performance Evaluation Measurement

This section introduces measurement techniques that were used to perform analytical performance evaluations of voice and data in integrated and nonintegrated networks. Three voice and data

communications networks were discussed in the mathematical analysis. The purpose of this analysis was to gather information on voice and data networks and use this information to create a simulation model. Using the created simulation model it was possible to study the different criteria and their effect on network performance. This section also gives a description of the method of analytical measurement followed by a brief discussion of three network topologies and the variables used in the actual measurement. Finally, a mathematical analysis using specified variables was discussed.

<u>Measurement Approach.</u> The overall measurement approach includes the analysis of delay, throughput, power, buffer capacity and cost in a network. The first and largest area to be addressed when discussing measurement approaches is delay time. Circuit switched delay and packet switched delay are discussed for voice and data networks. The integrated voice and data networks are analyzed both using priority traffic and using a first come first serve(FIFO) queuing scheme. The M/M/1 queueing theory is used in this study.

Prioritizing voice over data or data over voice requires a different measurement approach than FIFO. The analysis for determining delay time using prioritized voice and data is shown in two steps. The first step shows delay time equations for voice, priority (1), and data, priority (2), separately. The second step shows delay time for prioritized voice and data in a combined form. The priority scheme used in the following equations is subscript 1 for priority (1) and subscript 2 for priority (2). Priority queueing

time is given as follows(61:127)

$$E(w1)\text{seconds} = \frac{(\rho_1/\mu_1 C)+(\rho_2/\mu_2 C)}{1-\rho} \qquad (2.1)$$

The second priority queueing time is represented by

$$E(w2)\text{seconds} = \frac{(\rho_1/\mu_1 C)+(\rho_2/\mu_2 C)}{(1-\rho_2)(1-\rho_1-\rho_2)} \qquad (2.2)$$

The average waiting time of priority (1) messages is shorter than priority (2) messages. A priority (1) message enters the queue in front of all priority (2) messages except one which is already being processed. Considering the two priority classes with exponentially distributed message lengths the average wait time is (61:128)

$$E(W)\text{seconds} = \frac{(\rho_1/\mu_1 C)+(\rho_2/\mu_2 C)}{1-\rho} \qquad (2.3)$$

where

- ρ_1 = utilization of voice traffic percentage
- ρ_2 = utilization of data traffic percentage
- μ_1 = service rate of voice traffic in seconds/bit
- μ_2 = service rate of data traffic in seconds/bit
- C = channel capacity in bits per second
- w = denotes waiting time
- ρ = total utilization of the system

To find the average delay time of the entire system, the waiting time must be added to the average service time. Since the service rate for the study is the same for each priority then the average service time is determined by message transmission time ($1/\mu$) and channel capacity (C). The equation for average service time is (4:170)

$$\text{average service time } (\mu) = \frac{1}{\mu C} \tag{2.4}$$

Combining equations (2.3) and (2.4) gives the total average delay time in a prioritized network(61:127)

$$E(t) \text{ seconds} = E(W) + \frac{1}{\mu C} \tag{2.5}$$

where

t = denotes total delay time in system

When using this approach the time delay of higher priority messages is reduced at the expense of lower priority messages(61:128).

The approach is different when prioritization is not a factor. Average delay time is combined for voice and data using the FIFO approach. The average delay including service time at each node is (65:62)

$$d(i) \text{ seconds} = \frac{1}{\mu(i) * C - \lambda(i)} \tag{2.6}$$

In these equations

ρ = utilization in percentage
C = channel capacity in bits per second
μ = service rate in seconds per bit
λ = arrival rate in bits per second
i = number of the node

The Kermani and Kleinrock study (42) presents a similar equation for average node delay. The equation is (42:1053)

$$T \text{ seconds} = \frac{(Im + Ih)/C}{1 - \lambda(Im + Ih)/C} \qquad (2.7)$$

where

Im = message size in bits
Ih = header size in bits
C = channel capacity in bits per second
λ = arrival rate in bits per second of messages

The difference between equation (2.6) and (2.7) is that equation (2.7) accounts for a header containing the destination address of the message being transmitted. Kermani and Kleinrock(42) make the assumption that the service time equals the message size [$(1/\mu) = Im$]. Using the assumption [$1/\mu = Im + Ih$], equations (2.6) and (2.7) are equal. In order to find the average end to end delay for transmission of a message through a system, equation (2.7) is multiplied by the number of nodes (nh) in the messages path(42:1053)

$$T(tot) \text{ seconds} = \frac{(Im + Ih)/C}{1 - \lambda(Im + Ih)/C}\, nh \qquad (2.8)$$

where

nh = number of nodes in a message path
$T(tot)$ = total message delay

Equation (2.8) is used in this study for calculating the average delay for a message/packet switched communications networks.

The other half of packet delay is concerned with the reliability of the network. "One of the requirements usually imposed on computer networks is that they be reliable, even in the face of unreliable IMP's and lines(65:36)." A network must have redundant paths in order to attain this reliability. These redundant lines will not

always be the same length as the initial path, therefore presenting the possibility of lower performance(65:36). In order to reduce costs and delay time on a system the message path used should be the shortest path to the destination. Due to line and node failures, it was impractical to assume that all messages always take the shortest path to destination. To account for the use of these redundant lines a method to implement path percentages was used in this study. Adding an additional node simulates a redundant communications line. For the analytical model, routing through additional nodes was calculated into the overall delay. In order to obtain the effect of taking a redundant, longer path, a certain percentage of the entities to be evaluated were routed over the redundant path. In the mathematical analysis phase the probability of taking the shortest path is seventy five percent and the probability of taking the redundant path with one additional hop is twenty five percent. One out of every four packets was routed through the additional node and the delay time was added to its overall delay.

This method of evaluating the effect of a packet not taking the shortest path has its limitations. First of all, the redundant path was limited to only one additional node. This was due to the additional calculations required in mathematical analysis and the coding required in the simulation analysis for each node added to the network. The second limitation was the percentage of packets routed through the additional node. In an actual system the percentage of packets which traveled over the redundant path depends on the traffic intensity of the shortest path. Since there was no available means of determining these redundant path percentages an analysis of a wide

range of possibilities is necessary. These percentages were altered in the simulation analysis to observe the effects of packets which did not take the shortest path to destination.

The third type of delay involved the time to set up and send information over a complete end to end path. This type of delay is found in circuit switched networks. The time to transmit the channel reservation signal for a circuit switched channel is (42:1054)

$$Th \text{ seconds} = \frac{Ih * Nh}{C} nh \qquad (2.9)$$

This delay time is based on the number of node to node paths (nh) between start and destination of a message. Once the path has been reserved the message is transmitted in one hop fashion to its destination. The time to transmit the request for transmission signal, the channel release signal, and the message is given by the equation (42:1054)

$$Tr \text{ seconds} = \frac{2\ Ih * Nh + Im}{C} \qquad (2.10)$$

The two previous equations give the entire time required to send a message over the entire circuit to destination. These equations have the variables

Ih = header size in bits
Nh = number of channels available
C = channel capacity in bits per second
Im = message size in bits
nh = number of node to node paths
Th = delay time of the header
Tr = delay time of message transmission

The total delay for a circuit switched node is

$$Tcs \text{ (seconds)} = Th + Tr \quad (2.11),$$

where
Tcs = delay time of circuit switched node

Total delay time for a communications network is a combination of these delay equations depending on the particular network design used. The maximum delay time for networks with two separate circuits was the larger of the two average delay times. For example, in the network with the circuit switched voice model and packet switched data model the larger delay produced by these two models was the total overall delay for the network.

Throughput is an important measurement in a communications network. Average throughput is a ratio of the average number of messages or packets in the system to the average overall delay time of a system. The average number of messages in the system and the average system delay was calculated to determine throughput. The average number of messages or packets in the system was given by equation (4:170)

$$\bar{n} = \frac{\rho}{1 - \rho} \quad (2.12)$$

A stable environment throughput for a node is equal to the number of arrivals at that node. The equation for throughput is

$$Tput = \frac{\bar{n}}{\bar{d}} \quad (2.13)$$

where

$\bar{n}$ = the number of packets in the system

$\bar{d}$ = the average delay time of the system

Another equation used to check performance of the system is power. Power is a tool used by the manager to measure throughput and delay. The manager is concerned with throughput and the user with delay. Using power as a performance measurement allows the manager to improve his network performance and at the same time please the user(40:19). The equation is(40:19)

$$\text{power} = \text{throughput/total delay} \tag{2.14}$$

Buffer capacity is another important measurement because of the effects it has on delay, cost and availability. In this study buffer capacity was determined by the probability of rejection of a packet. According to Forgie and Nemeth, if the probability of packet loss is kept small, on the order of one percent, then users will find voice communications to be acceptable(23:38.2-44). The probability of rejection was specified by allowing only a specific number (K) messages to be waiting in a queue at a given time. A reasonable buffer length (K) such that (4:170)

$$P[\text{ K or more waiting }] = P[\text{ K+1 in the system }]$$

$$\leq \text{Probability of message rejection}$$

$$\rho^{(K+1)} \leq \text{Probability of message rejection(PMR)}$$

$$\frac{\ln \text{PMR}}{\ln \rho} - 1 \leq K \tag{2.15}$$

The final measurement of this study was cost. The three areas of cost measurement was switching cost, tariff (mileage) costs and voice digitization cost. Switching cost was determined using the equation(14:352-353)

$$Sc = C*Nc + P*Np + H*Nh \qquad (2.16)$$

where

C*Nc = circuit cost * number of circuit switches
P*Np = packet cost * number of packet switches
H*Nh = hybrid cost * number of hybrid switches

Tariff (mileage) cost is a set rate cost plus costs based on distances between nodes. Voice digitization cost was based on the a set rate for each voice digitization rate (VDR). Overall cost was the summation of the three types of cost(31:1555).

Total cost = Switching cost + Tariff cost + VDR cost (2.16)

Accuracy in calculating cost was very difficult due to rapidly changing technology and costs(31:1552). This analysis gave a general idea of cost comparison between different switching techniques, distances and speech encoding rates.

Performance Approaches. This portion of the analysis discusses three network designs. The three network topologies are, a network configuration with separate communication paths for voice and data; a network with voice and data queued over the same path; and a network configured with voice and interactive data over one path and bulk data over another path. The first network topology uses separate communication paths for voice and data. Voice uses a circuit

switched network design for digital voice traffic and data uses a packet switched network design for digital traffic.

A circuit switched network establishes a complete end to end communications path from source to destination prior to transmitting a message. A channel reservation signal travels node by node through a network reserving the communication path as it goes. The channel reservation signal queues at each node until the node is available, this is called queueing delay. Once the signal enters the node it is serviced,the node added to the reserve path and the packet sent to the next node. The time to be serviced is the service time. When the channel reservation signal reaches the desired destination the path is complete and a request for transmit signal is sent back to the source node. Since the path is already reserved this request for transmit signal is only affected by a single servicing. The request for transmit signal travels from destination to source node in a single hop being serviced one time with no queueing delay. When the signal reaches the source node the message is transmitted in the same fashion as the request for transmit signal except in the opposite direction. The message is not affected by queueing delay, because the channel was reserved. Once the message is received at the destination node a channel release signal is sent along the path to free the communications path for other traffic. The channel release signal, like the request for transmission signal and the message transmission signal, is only serviced once. Once the channel is released one message transmission is completed. A circuit switched model is used to represent the circuit switched network for voice(see Figure 1).

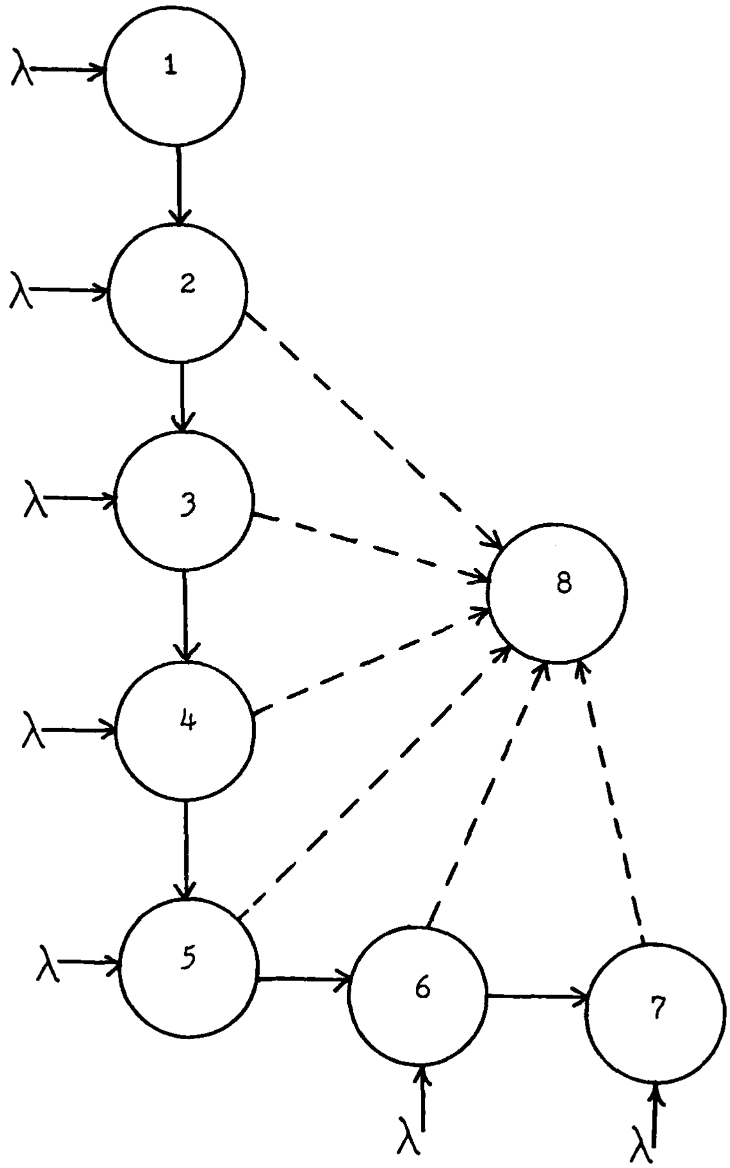

Figure 1. Circuit Switched Model for Voice

Nodes 1 through 7 represent actual network nodes with (λ) arrivals at each node. The destination of the message determines where node 8 (output) is connected. For example, if the destination of a message originating at node 1 is node 3 then the output node, annotated as node 8, is connected to node 3. Node 8 is not a network node but an output node to collect statistics on network performance.

An analysis of this circuit includes checking the output node 8 at five possible destinations.

The packet switched network transmits packets of voice and or data node by node over a communications path. Voice and data messages are broken down into specified packet sizes and transmitted through the network and reassembled at the destination. These packets have headers which carry the destination address. Packets are received and serviced by each node. Once the next node becomes available the packet is sent on until the packet reaches its destination. The first network topology uses a packet switched model for sending all data(see Figure 2).

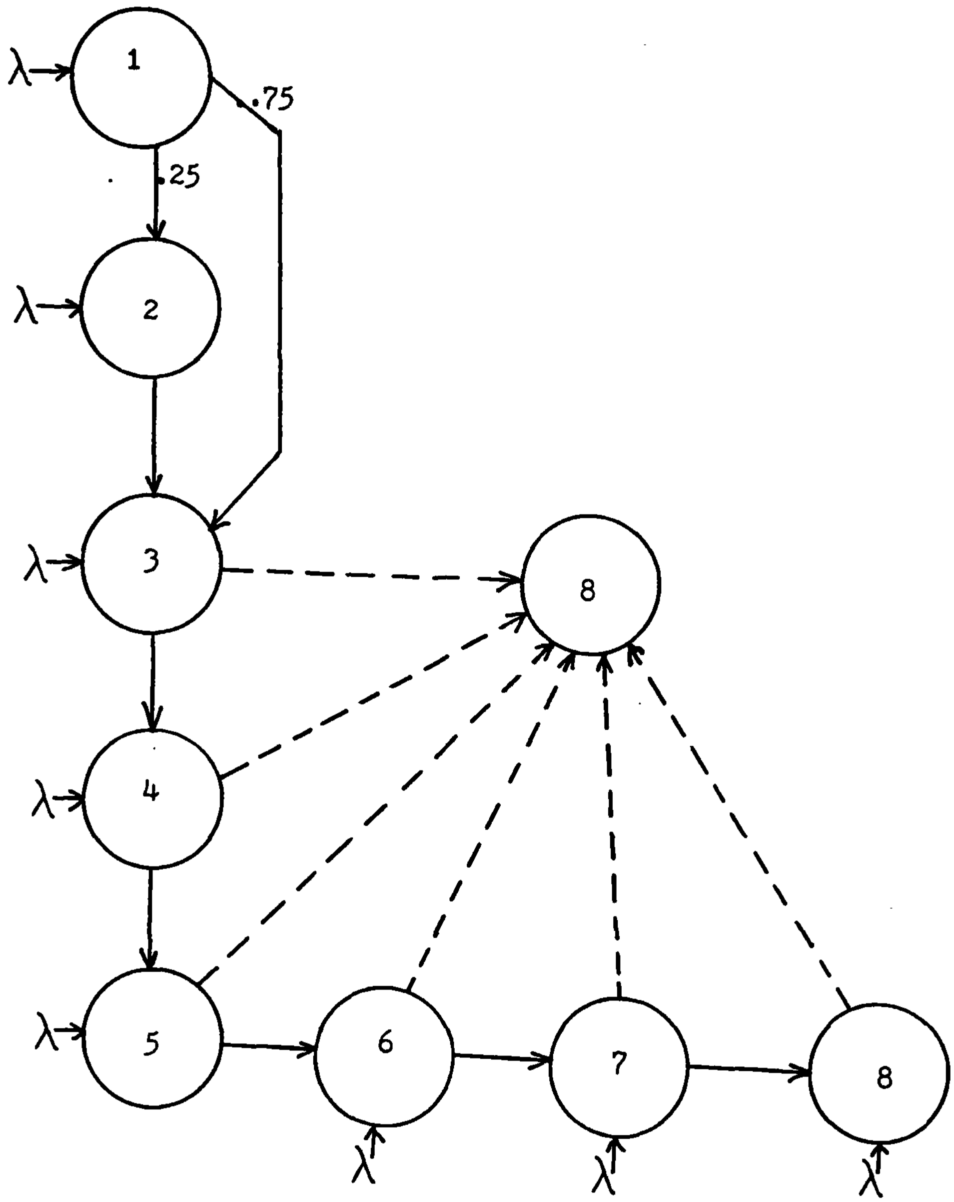

Figure 2. Packet Switched Model for Data

The design of this circuit has 8 network nodes with arrivals (λ). Node 2 is an alternate node for routing twenty five percent of the traffic over a redundant path with one additional node. The rest of the traffic took the shortest path to destination skipping node 2. Nodes 1, 3, 4, 5, 6, 7, and 8 are primary nodes, the shortest path to destination. The packet switched model used output node 9 as a

logical node to gather statistics on the networks performance.

The second network topology used a packet switched model to represent a packet switched network for transmitting digitized voice and data over the same path(see Figure 3). The packet switched model shown in Figure 3 performs the same as the packet switched model of Figure 2 except it has three arrivals.

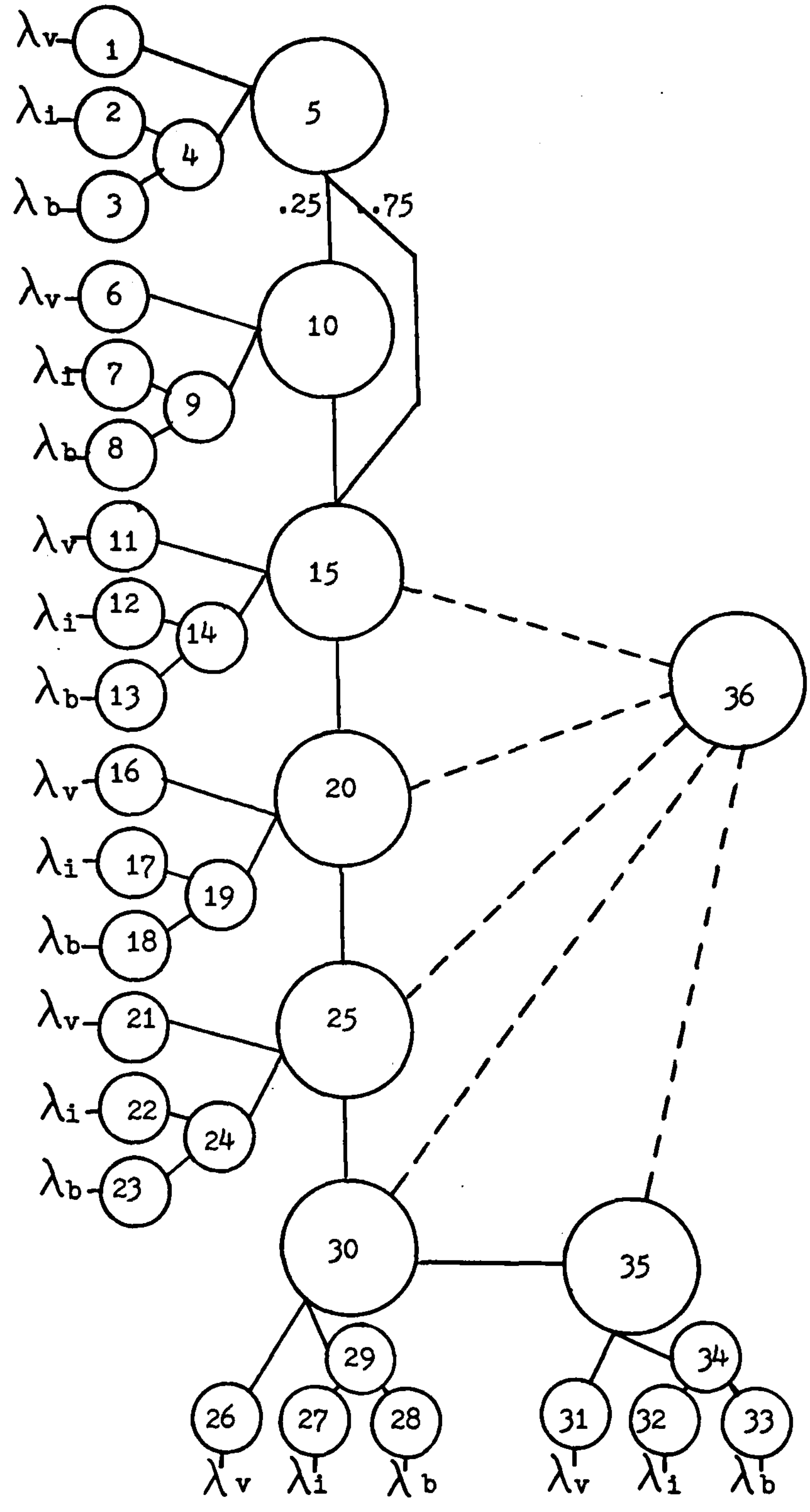

Figure 3. Packet Switched Digital Voice and Data

The network design shown in Figure 3 integrates voice, interactive data and bulk data over the same circuit. Voice is digitized for this network. Interactive and bulk data are separated because of the differences in message size. This design allows voice to be prioritzed over data to prevent excessive voice delay or allow a FIFO scheme to be used. The network nodes are 5,10,15,20,25,30 and 35. Node 10 is an alternate node for routing the percentage of traffic which travels an additional node to its destination. Messages which take the shortest path to destination skip node 10. This network design permits the percentage of voice to data and interactive to bulk data to be changed to evaluate the effect these percentages have on network performance. Arrival rates for voice, interactive data and bulk data are noted on the figure by, λ v, λ i and λ b, respectively. Node 36 is an output node for collecting statistical information on network performance. This logical node can be connected to any network node to check network performance at a particular location.

The third network is a combination of the first two networks. A packet switched network model is used for bulk data and a packet switched network model is used for combined digital voice and interactive data. The packet switched model used for bulk data is identical to the packet switched model found in Figure 2(see Figure 4).

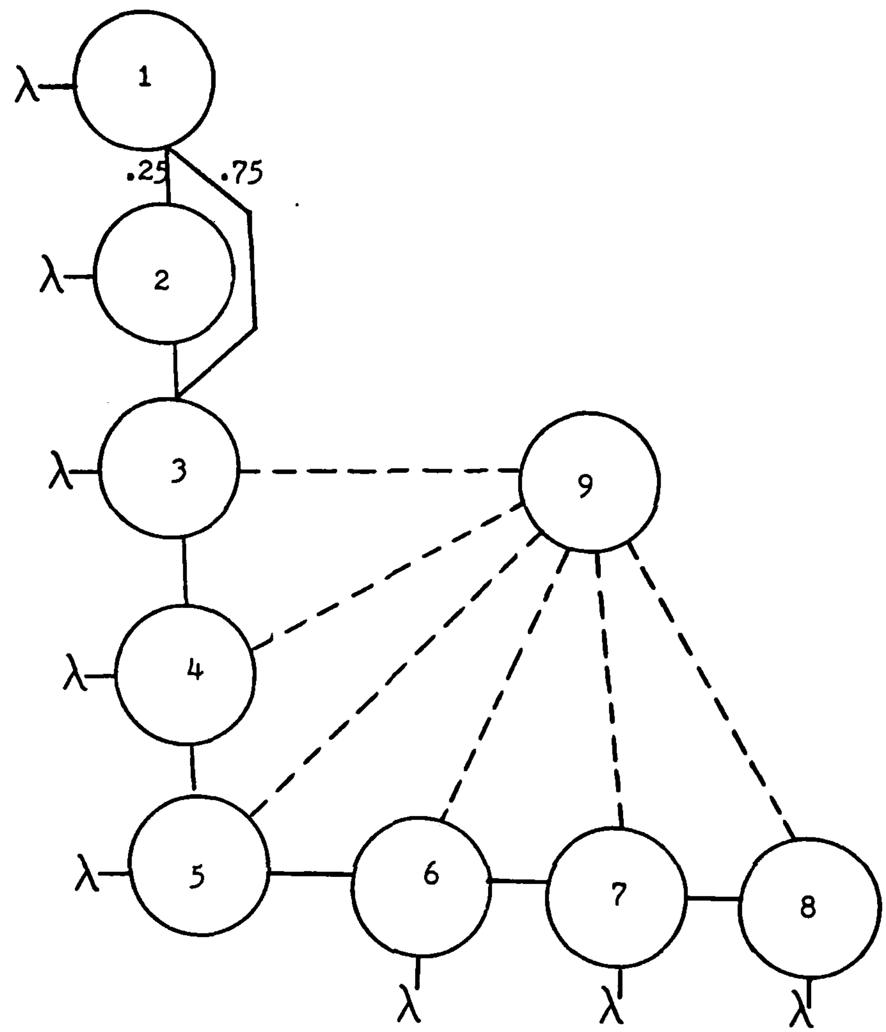

Figure 4. Packet Switched Model for Bulk Data

This packet switched design uses the same routing and output capabilities as Figure 2. The difference is the arrival rate to each node.

A voice and data integrated topology similiar to Figure 3 is used for the voice and interactive data of the third network design(see Figure 5).

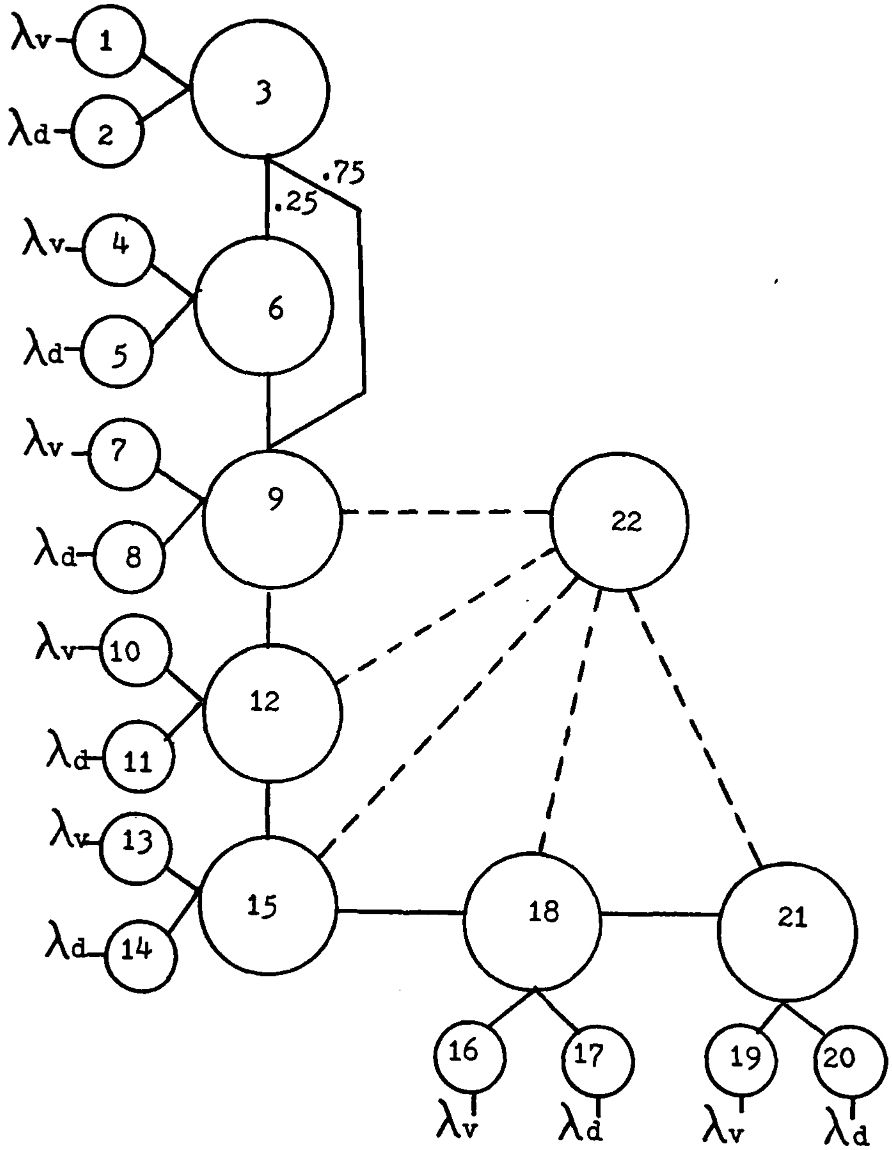

Figure 5. Packet Switched Model for Voice and Interactive Data

This design integrates digitized voice and interactive data over the

same circuit. The network nodes are 3,6,9,12,15,18 and 20. Node 6 is the alternate path node used to route the percentage of traffic which uses on additional node to get to its destination. Shortest path messages go from node 3 to node 9. This network allows prioritization of voice over data as well as the capability to alter the percentages of voice and interactive data through the network. This was done from the logical nodes which enter each network node. Arrival rates which enter the logical nodes for voice and interactive data are denoted by λ v and λ i, respectively. Node 22 is the output node for collecting statistics on network performance. This logical node was used to check network performance at each network node. The percentages of voice versus data was altered to determine the effect on network performance. The same path routing and output capabilities as in previous designs were used in this model.

All three network topologies represent actual or possible network implementations. The first network simulates present day separated analog voice and digital data systems. The model shown in Figure 1 represents the analog voice circuit switched network. The packet switched model of Figure 2 is modeled after the digital data network know as the DDN. The second and third network models simulate alternative approaches to future voice and data communication networks. These simulation models were used to analyze various performance criteria to determine the best approach for voice and data communications

Performance Variables. Performance variables for this analysis were taken from research studies of the Defense Data Network(13,19),

personal interviews(50,59), and DOD communications research(30,31). These particular variables are used in order to verify and validate the simulation models. Performance variables were broken down into five major categories. These categories are packet information, arrival rates, service rates, voice information, percentage variables and cost.

The first category of performance variables are message and packet information. The average voice message size used in this study was assumed to be packet length. The message/packet size was set to different sizes and therefore changing the service rates using equations (2.9) and (2.10). This assumption which sets voice message size equal to packet length was necessary for comparison of the different networks. Data traffic maintained a 40 bit message size for interactive data and a 23,200 bit message size for bulk data(59). The large message size of bulk data does not have a detrimental effect on the arrival rate and service rate because only 6.3 percent of messages were bulk data with 93.7 percent interactive data messages(19). The packet size selected for this data is 2048 bits(59). This packet carries 1920 bits of message information and 128 bits of header.

Arrival rates for the simulation models were calculated using the message/packet information and the average arrival rate of messages into the DDN. The average DDN arrival rate is 20 packets per second(50). Since two of the simulation models use separate arrivals for interactive and bulk data it is important to determine these arrival rates. Interactive data messages average 40 bits in length which was less than packet information size. Since interactive data

messages were less than packet size, packet size was used for interactive data arrival rate calculations. Bulk data message size was converted to packets by dividing 23,200 bits by 1920 bits, which gives a message size of 12.08 packets. For routing purposes a control packet was required for each message packet being transmitted. Multiplying the message size in packets by two, to account for control packets, and then multiplying by the percentage of interactive and bulk messages arrivals gave the number of packets arriving for a given a time period. In a given a time period 152.208 packets of bulk data and 187.4 packets of interactive data arrive for a total of 339.608 packets. Using an arrival rate of 20 packets per second, the DDN average for data, and multiplying it by the ratio of interactive and bulk packets to the total number of packets gave arrival rates for interactive and bulk data. The arrival rate for interactive was 11.04 packets per second and the arrival rate for bulk data was 8.96 packets per second. The arrival rate of voice was assumed to be equal to the total number arrivals of data, 20 packets per second.

Service rates were determined based on the average service rates of the DDN and the calculated service rates on the circuit switched model. The average service rate used for all packet switched nodes was 333 packets per second(50). Circuit switched service rates for network model shown in Figure 1 was calculated based on equations (2.9) and (2.10). The service rate for the channel reservation signal was .0023 seconds per packets from equation (2.9). The service rate for the message transmission, request for transmission signal, and channel release signal was .0388 seconds per packet from

equation (2.10). The service rate for the message transmission and channel release signal was based on messages of one packet length. For messages greater than packet length the service rate changes. Both circuit switched service rates change based on packet and header size.

Voice information included a discussion of voice digitization rates and the percentage of voice versus data. Voice digitization rates were based on available channel capacity. Some voice digitization rates are 2.4 kilo-bits/second, 16 kilo-bits/second, 32 kilo-bits/second, and 64 kilo-bits/second. For this study voice digitization rates were assumed to be equal to 56 Kbps, the backbone channel capacity for the DDN. The voice digitization rate can be altered by changing the number of servers at each node of the simulation model. Also important to the analysis of voice was the percentage of voice and data transmitted over a channel. In a study by Calabrese(8), the highest percentage of voice over data was thirty percent which was extremely low based on present day communications. Comparison of present traffic needs show the voice traffic load was greater than data and possibly be as high as 80 to 95 percent(2:59). This analysis used percentages which range from ninety percent voice and ten percent data to ten percent voice and ninety percent data. Prioritization of voice and data was also important to this study. The analysis included delay time calculated with and without voice prioritized over data. The channel capacity used 56 kilo-bits/second, the backbone channel capacity for the DDN.

The percentage of bulk versus interactive data, the percentage of path algorithm and the probability of rejection was necessary to

evaluate the effect changes in the quantities of voice and data have on network performance. Using percentages to control the flow of interactive and bulk data allowed one to be prioritized over the other. In this research the percentage of interactive versus bulk data ranged from 90 percent interactive and 10 percent bulk to 10 percent interactive and 90 percent bulk. The percentage for path algorithm was used to see the effect of path routing on network performance. The ratio for shortest path and one additional hop path was 75 percent shortest path and 25 percent longest path. The third area of analysis using percentage variables is probability of rejection. The probability of message rejection is the percentage of messages or packets lost given a specific buffer capacity. According to the Forgie and Nemeth study (23) a probability of rejection of one percent is within acceptable standards for the user. These percentage variables provide insight to changes of network message flow which allow the user to select the best approach based on given requirements.

The final category of performance variables, cost, was divided into three sections. The three sections were switching cost, tariff(mileage) cost and voice digitization cost. Switching costs were based on a ratio of cost to channel capacity taken from the Chou study(14). The circuit switch ratio is $750K/200Kbps. The packet switch ratio is $125K/100Kbps. The hybrid switch ratio is $170K/240Kbps. Tariff(mileage) costs were based on a telephone interview with Prishavalco(50). This cost starts with a base rate of 28,745 dollars annually. Mileage cost was calculated using Table 2.

TABLE 2

TARIFF(MILEAGE) COSTS

Cost	Mileage
$ 155.40	1 - 15 mi
$ 129.60	15 - 25 mi
$ 97.20	25 - 100 mi
$ 56.40	100 - 1000mi
$ 34.80	1000 - up mi

Distances were calculated based on possible locations in the DDN(50)(see Figure 6).

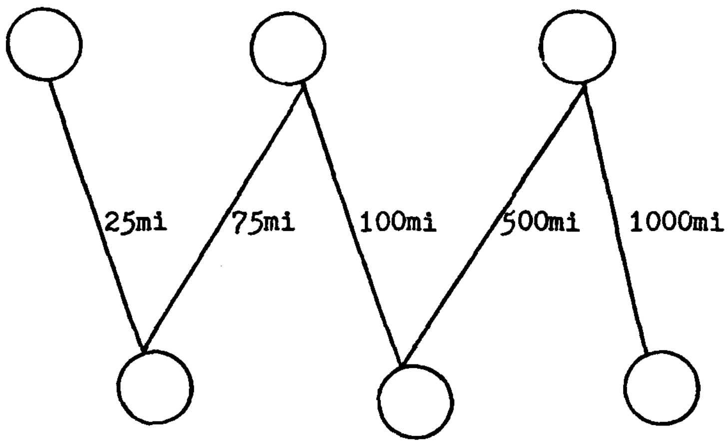

Figure 6. Node milages

The costs are taken from a discussion with Mr. John Salerno(59). The cost rate for digital voice transmitted over a circuit switched path is approximately 350 thousand dollars per node. The packet switched node for data transmission is 40 thousand dollars per node. In order to send digital voice over a packet switched network the cost is increased by 400 thousand dollars per node to account for digital

voice and "silent period detection". The hybrid switched node is a sum of the circuit switched costs for voice and the packet switched costs for data, totalling 390 thousand dollars.

Mathematical Analysis

The mathematical model was an analytical tool used to verify the simulation model results. This analysis was based on the equations, specific performance criteria, and the three network topologies previously discussed in this chapter. The results which were of the most importance in this analysis were delay time, throughput, power and cost. Since throughput and power were determined using delay time then the criterion delay time was a key factor for verification of the simulation models. Specific constants were necessary in order to evaluate delay time in the mathematical analysis and the simulation models. The constants for the mathematical and simulation models were traffic intensity, percentage of path algorithm, arrival rates and service rates. This section discusses the mathematical analysis of the results which was used for the verification of the simulation models.

The first constant to evaluate in the mathematical analysis was traffic intensity. It was important to maintain a constant traffic intensity in order to evaluate the performance of a network with respect to the time a message takes to travel from source to destination. Each of the three given networks consist of six permanent nodes. The packet switched network included a seventh node which was the node used for alternate path routing. The alternate

node was never used as a possible destination. If a message was routed through the network it was equally likely the message terminated at any of the six possible permanent destinations. In the same respect, arrivals at node 2 terminated equally at each of the next five destinations and so forth. In order to maintain a constant traffic intensity only 5/6 of message arriving at the second permanent node were transmitted to the third permanent node. Similarly, only 4/5 of messages arriving at node 3 were transmitted to node 4, 3/4 of arriving messages at node 4 were transmitted to node 5, 2/3 of arriving messages at node 5 were transmitted to node 6, and 1/2 of arriving messages at node 6 were output. Since external messages arrived at the same rate to all nodes this method was necessary to maintain a constant traffic intensity.

It was necessary to determine utilization for a network in order to find maximum queue length using the rejection probability. A value of $\rho = .75$ was randomly assumed to determine maximum queue lengths. The use of the percentage of arrivals output at each node and an initial $\rho = .75$ gave the external traffic intensities in Table 3. Using a probability of rejection of one percent, specified traffic intensities and equation (2.13) the queue length at a particular node was determined. The largest possible queue length allowed at each node to maintain less than a one percent possibility of rejection is specified in Table 3. Since all packet switched models used the same values of ρ, then the maximum queue length was the same.

TABLE 3

EXTERNAL TRAFFIC INTENSITY AND QUEUE LENGTH

a) External Traffic Intensities for Circuit Switched Design

Node#:	1	2	3	4	5	6
ρ :	.75	.125	.15	.1875	.25	.375

b) External Traffic Intensities for Packet Switched Design

Node#:	1	2	3	4	5	6	7
ρ :	.5625	.75	.125	.15	.1875	.25	.375

c) Queue Lengths for each Node

Node#:	1	2	3	4	5	6	7
Queue Capacity at K=1%	16	16	2	2	2	3	4

The path algorithm percentage was based on the number of messages which took the shortest path to destination versus the percentage of messages which took an additional hop enroute to destination. The probability of messages taking the shortest path depended on availability of nodes. For this study it was assumed that only 75 percent of all packets take the shortest path to destination. The other 25 percent of messages go through the alternate node enroute to destination. The path algorithm was only significant to the packet switched networks for this analysis.

The first network to be analyzed is shown in Figure 1 and 2. This network consists of a circuit switched model for voice and a packet switched model for data. The circuit switched model used

equations (2.9) and (2.11) to calculate average delay time required for a message to reach destination node 6. The external arrival rates for each node was 20 packets per second. This arrival rate is based on the assumption that the message size was 1920 bits and the header was 128 bits as described in the performance criteria section. Using equation (2.9) the calculated delay time was the time it takes to reserve each individual node to node link, .0022857 seconds. Multiplying this time by 6 gives .0136143 seconds, the time to reserve the entire path. The time to request transmission, transmit the message and release the communication path was .0388571 seconds using equation (2.11). The total time to transmit a message over the circuit switched model from node 1 to node 6 was .0547571 seconds. The data traffic had an arrival rate of 20 packets per second and a service rate of 333 packets per second at each node. Using queueing matrix analysis the average delay time was calculated by determining the new arrivals at each node and the average node delay using equation (2.6). Summing the node delay times for all nodes including the alternate node gave a average delay time of .0232126 seconds for a packet to travel from node 1 through node 7. The arrivals, service rates, and average delay times for the first network are displayed in Table 3.

The second network consists of a single packet switched model with three arrivals as shown in Figure 3. The arrival rates were 20 packets per second for voice, 11.04 packets per second for interactive data and 8.96 packets per second for bulk data. The delay time for this packet switched model was calculated using the same method as the packet switched model of the first network. Using

queueing matrix analysis and equation (2.6) the average delay time was calculated for the second network for the path node 1 through node 7. The average delay for the second network using a service rate of 333 packets per second was .0260656 seconds. The average delay for the second network is displayed in Table 3.

The third network is a combination of two packet switched models as shown in Figure 4 and 5. The first packet switched model of Figure 4 was for bulk data and had an arrival rate of 8.96 packets per second with a service rate of 333 packets per second. The average delay time for the path from node 1 through node 7 was .0219396 seconds. The second model of the third network serviced digitized voice and interactive data as shown in Figure 5. The arrival rates for the second model were 20 packets per second for voice and 11.04 packets per second for interactive data. Using queueing matrix analysis the average delay time for the path node 1 through node 7 was .0246831 seconds. The performance statistics for this network is also shown in Table 4.

TABLE 4

MATHEMATICAL MODEL RESULTS WITHOUT PRIORITIES

Network Type (number)	Arrival Rate (pkt/sec)	Service Rate (pkt/sec)	Delay Time (sec)	Through-put (pkt/sec)
1				
Circuit Switched				
Voice Path	20	434.78	.0136143	
		25.77	.0388571	
(TOTAL)			.0547571	
Packet Switched				
Data Path	20	333	.0232126	3.35
2				
Packet Switched				
(Combined Path)				
Digital Voice	20			
Interactive Data	11.4	333	.0260656	3.85
Bulk Data	8.96			
3				
Packet Switched				
(Combined Path)				
Digital Voice	20	333	.0246831	3.45
Interactive Data	11.04			
Packet Switched				
Data Path	8.96	333	.0219396	3.15

Priorization of voice over data changes the approach to determining delay time. Delay with priority to one type of arrival over another was calculated using equations (2.3) and (2.5). In order to find voice utilization and data utilization it was necessary to use the equation $\rho = \lambda / \mu$, the individual arrival rates for voice and data, and the overall from Table 3. The prioritized delay time for networks 2 and 3 are shown in Table 5.

TABLE 5

MATHEMATICAL MODEL RESULTS WITH PRIORITIES

Network Type (number)	Service Rate (pkt/sec)	Arrival Rate (pkt/sec)		Delay Time (sec)
		Voice	Data	
2				
Packet Switched (combined voice interactive data and bulk data)	333	20	20	.02562680
Packet Switched (combined voice and interactive data)	333	20	11.04	.0245951

The cost analysis provided a general idea of comparison of costs of the different models. The costs analysis is shown in Table 6.

TABLE 6

MODEL COSTS

Network Type (number)	Mileage Cost (millions of dollars)	Voice Digitization Rate(millions of dollars)	Switching Cost (millions of dollars)
1			
Circuit Switched Voice	.1208	2.10	1.26
Packet Switched Data	.1208	.240	.4200
2			
Packet Switched (combined digital voice, interactive data and bulk data)	.1208	2.64	.4200
3			
Packet Switched (combined digital voice, and interactive data)	.1208	2.64	.4200
Packet Switched bulk data	.1208	.240	.4200

Mileage results were based on actual mileage costs from Table 2 and assumed mileages from Figure 5. The voice digitization costs were determined for 56Kbps using information from the Gitman(30) study. Switching costs were calculated using ratios of costs to channel capacity used in the Chou(14) study.

Discussion and Results

In the mathematical analysis the average delay time and throughput results show the third network with the lowest delay time, however the second packet switched path used for bulk data had very poor utilization. Comparing the packet switched model for bulk data of the first network and the packet switched model for bulk data of the third network show very little difference in delay and throughput but a large difference in arrival rates indicating a great deal of idle time in the bulk data model of the third network. The packet switched network of the second network indicated a better use of resources by comparing results and arrival rates with the two bulk data packet switched models which used lower arrival rates. Priorization was not a deciding factor between the second and third network because there was only one milli-second difference between their delay times. The significance of the priority analysis was in the fact that digitized voice was prioritized over data without a detrimental effect on the average packet delay time. This factor makes digitized voice networks comparable to networks which have single circuit switched paths for voice, as shown in the circuit switched model of the first network.

The cost analysis shows savings in maintaining separate circuit switched voice and packet switched data over those networks using digitized voice. The determining factor for digitized voice is the long term impact of delay time, throughput and efficient use of resources. Since results of delay time and throughput are similar when comparing the use of analog and digitized voice the network which used its resources more efficiently was the best choice of a communications network. The second network which integrated digitized voice and data over a single path offered the better overall performance results.

Conclusion

These results had significant impact on the following chapters. The key significance of the mathematical analysis results was for verification and validation of simulation models. The results of the mathematical analysis was used for validation because the results were based on inputs from actual communication networks. The results were also used to verify the simulation models. Similar results in the simulation model to that of the mathematical analysis indicated verification of the simulation models.

III. SIMULATION MODELS

Introduction

This chapter discussed the simulation language selection, performance criteria, the design of three voice and data network simulation models, simulation model modifications and the basic design of the Simplified Voice Trunking Model(SVTM). The simulation language must be selected based on availability of a language compiler, ease of use and knowledge of the author. Performance criteria were specified in chapter II, however some adjustments need to be made to adapt the criteria to the simulation models. The network models simulated were the same as those discussed in chapter II, Figure 1 through 3.

The first simulated network is comprised of two separate models, a circuit switched model for voice and a packet switched model for combined data. The second simulated network is a single simulation model with voice, interactive data and bulk data arrivals using the same path. Network three consists of two packet switched models. The first model sends voice and interactive data over the same path. The second simulation model is used for transmiting bulk data. Simulation model modifications are additions or improvements to allow more accurate flow control and output results. These simulation models were created in order to analyze possible approaches to voice and data communications.

Simulation Language Selection

An initial step to take when analyzing with the use of a simulation model is the selection of the appropriate programming language. There are two categories of programming languages available for use in a simulation study. The first category includes simulation languages such as GPSS, GASP, SLAM, and Simscript. The second category contains several languages like fortran, pascal or C. The first category provides a very high level language and frees the user from being required to know specific details about the operating system as does the second category(63:117). Simulation frees the analyst from repetitive work as that found in using mathematical analysis and affords more time to concentrate on results(63:93). A simulation language like SLAM uses an assortment of nodes and branches for modeling the flow of entities through a system(52:79). Other features of simulation languages provide continuous monitoring capabilities, initialization capabilities and an overall flexibility of design. The use of a simulation language provides the fastest and simplest approach for the creation of a simulation model.

The use of a general purpose programming language of category two also has its advantages. The language provides greater run time efficiency and are normally more available than simulation languages(26:14). The reluctance to learn a new language is not uncommon among programmers when selecting a simulation language for a specific study. General languages are however limited by increased coding and debugging difficulty.

A simulation language was chosen for this simulation study

because of the ease of implementation of the language and the specific simulation language selected, SLAM, provides excellent building blocks for this particular study. The largest disadvantage of using SLAM was in learning the SLAM language. The advantages were the availability of a SLAM compiler and the ease with which SLAM can be applied to network studies.

Simulation Model Design

This section discusses the performance criteria used in the simulation models, the network models and modifications to the models. The actual performance parameters used in the simulations for comparison with the mathematical analysis are designated in this section. The control and flow of the three network models presented in chapter II are discussed in this section. Modifications provide better control of entities passing through the models. This section explains the modifications made on the simulation models to improve control, alter flow and to prioritize entities.

Performance Criteria. Performance criteria were described in detail in chapter II, however several adjustments were required in order for their use in simulation models. The performance criteria used in the simulation model were arrival rates, service rates, source designation, and destination node. The arrival rates were the inputs to the simulation model and the latter were designated as attributes. The circuit switched and packet switched model attributes differ only slightly. Other criteria which were specified in this section were the percentage of voice over data, probability

of rejection statistics and the percentage of bulk data to interactive data.

The arrival rate used in the simulation models was converted to mean interarrival times for use in the SLAM simulations. Voice and combined data arrivals of 20 packets per second were inverted to .05 seconds per packet. Separate bulk data arrivals of 8.96 packets per second were inverted to .112 seconds per packet and 11.04 packets per second of interactive data were inverted to .091 seconds per packet.

The attributes for the SLAM simulation models depended on the type of model. There were three attributes used for the packet switched models. Attribute one was used to direct entities through the network to the designated destination node. In this simulation the destination node was node 7. Attribute two specified the service rate for a packet to be processed at a node. In this simulation study the service time of 333 packets per second was converted to a mean service rate of .003 seconds per packet. Attribute three designated the source node. Each permanent node designated itself as a source node. This attribute allowed the destination node to determine which distance was desired for measurement.

The circuit switched simulation model attributes differed from the packet switched model only with regard to the servicing of messages and final destination. Attribute one of the circuit switched model performed the same function as attribute one of the packet switched model and was designated as node 6. Attribute two provided the service rate at each node for the channel reservation signal. This value was already calculated in mean service rate form in chapter II. Attribute two was.0023 seconds per packet. Attribute

three was the service rate for the request for transmission signal, message transmission, and channel release signal. This value was also calculated in mean service rate form in chapter II to be .0333 seconds per packet. Attribute four was the source designation and performed the same as attribute three of the packet switched model. The attributes used in both models were used to control and service the flow of entities from input to output.

There were several other performance criteria used to control the simulation model results. The changing of the percentage of voice to data in the system allowed the user to alter the weight of traffic flow of voice and data in the network. There was no breakdown in percentage used in the initial simulation model. The initial percentage rates were necessary to maintain consistency between the simulation models and the mathematical analysis for verification and validation. Another criterion which affected network performance was the probability of rejection. The rejection probability in effect set the queue limits at each node so that packet rejection did not exceed specified limits described in Table 3 of chapter II. The probability of rejection for the simulation models was one percent. The percentage of bulk to interactive data was another criterion for controlling simulation results. This criterion allowed the weight of traffic flow for interactive data and bulk data to be altered. The percentage of bulk to interactive data was strictly based on arrival rates. The initial performance criteria was established in order to verify the simulation models with the mathematical analysis. A network analysis was accomplished using the ranges of performance criteria values found in chapter II.

Network Model I. The first network was made up of two separate simulations. The first simulation represents circuit switched voice and the second simulation represents packet switched data. The circuit switched voice simulation was modeled in SLAM using a queueing network design. Messages arrived at the same exponential rate to each node beginning at time zero. At each node attributes were assigned the number of the source node, the service rate for the reservation signal, the service rate for the message transmission, and the destination node number. The first node serviced the channel reservation signal and sent it to the next node. The next node serviced the channel reservation signal and checked attribute (4) to see if the desired destination had been reached. If the desired destination had not been reached the channel reservation signal was sent to the next node and followed the same procedures till reaching destination. This procedure reserved the entire end to end path for message transmission. If the channel reservation signal had reached its appropriate destination then the request for transmission signal, message transmission and channel release were serviced in a one hop fashion based on attribute (3). The only entities serviced were those with the appropriate source designation in attribute(4). This process was repeated over a specified time span to determine the average time to send the message through the model. These results were collected using the COLCT node which calculated the average time for an entity to pass through the network.

The packet switched data simulation also used a queueing network design with packets arriving at an exponential rate to each node. At the first node the packet was serviced and transmitted to the next

node. If the next node was unavailable the packet waits in a queue for access. When the packet enters the node the packet was serviced according to the service rate specified in attribute (2). The destination attributes (1) was checked to determine if the packet had reached its destination. If the packet had not reached its destination it continued through the network in the same manner. If the packet had reached its destination it was sent to the output node where statistics on packet system time were collected. The packet was sent to a COLCT node and the time for a packet to process through the network was measured and averaged over a specified period of time. Also included in the packet switched simulation was an alternate path routing. Specific percentages of packets took an alternate route which added an additional hop onto its path length.

Network Model II. The second network model, which integrates voice interactive data and bulk data over a single path, is very similar to the packet switched simulation of the first network model. The differences are in arrivals to each node. Using this model the user can alter percentages of voice over data, and percentages of interactive data over bulk data to determine ranges of the simulation model and the performance parameters. This simulation provided an average delay time for all packets in the system by using the COLCT node to collect statistics.

Network Model III. The third model is split into two separate simulations. The first simulation combined voice and interactive data over a single packet switched path. The second simulation is a single packet switched path for bulk data. The packet switched

simulation model for the bulk traffic of the third network is identical to the bulk data simulation model of the first network except for the different arrival rates. The third network scheme represents another approach to voice and data communications by providing a separate path for bulk data and an integrated path for voice and interactive data. This model also allowed for use of percentages to control voice and data over the system. In each case a COLCT node collected average delay time for designated packets.

Modifications. Modifications to the three network models increased the capability of the simulations. Modifications included Blocking, Balking and Prioritization. The first modification provided a method to calculate packet loss on the network. Using the queue lengths specified in Table 3 for one percent probability of rejection blocking was used to maintain queues at maximum queue length. Blocking at a node only allows a specific number of entities(packets) to queue. This allowed the designer the capability to control queues at critical nodes. Balking is a feature placed on a node which precedes a blocking node. If an entity is blocked from entry into a queue then it becomes a loss. The use of Balking allows the designer means to check the percentage of packets rejected so changes can be made to limit their occurance. At each input node a balking procedure was used to collect voice and data packets which were blocked out of the systems.

Prioritization was a modification which permitted one entity to have priority over another entity either in the queue or in the entire node. The use of a low value first(LVF) priority at a

specific node allowed entities with a lower mean interarrival rate value to have priority over the higher valued entities. The high value first(HVF) denoted just the opposite functioning than the (LVF). Another characteristic of prioritization was [NCLGR, HVF(JEVNT)], a secondary priority scheme that prevented the interruption of a packet being processed. Omitting the secondary priority scheme allowed the prioritized entity to override a nonprioritized entity already being processed.

Validation and Verification

When creating the simulation model it was important to verify and validate simulation model results. Verfication of a simulation model compared simulation model results to expected results. Verification was accomplished when the simulation model results were reasonably close to desired output. Validation established the simulation model performance to be similar to that of a real system(52:10). Verification and validation can best be checked by comparing simulation output to an existing system output. This method of verification and validation was limited because the models were of a more general and hypothetical nature than those in existence.

The method used to validate the simulation models was by a comparison of input to output. This was somewhat difficult to do because of the multiple inputs and the percentage of traffic permitted to flow through the network. Validation of the circuit switched model and the packet switched model was accomplished in the same manner. Using the simulation time and the number of entities

which were processed through the models, output was compared to arrivals into the models. If the flow through the network is as designed then the model is representative of an actual system. This method was used to validate the simulation models. Analyzing the circuit switched model with 263 entities output in 100 seconds gave a throughput of 2.63 packets per second. The model was only examining input into node 1. It was assumed 1/6 of node 1 input of 20 packets per second was equally distributed among the six nodes, therefore 1/6 of 20 packets per second equal 3.33 packets per second. Using the value of ρ = .75 the total network output from the mathematical analysis was equal to 2.5 packets per second. The simulation model provided similar results to the mathematical model and was therefore validated.

Verification of the simulation models were accomplished by a comparison to a mathematical model of an actual system. The comparison insured that the flow through the system was representative of an actual system. Two methods were used in the mathematical analysis. The circuit switched model was analyzed using the approach of Kermani and Kleinrock(42). Queueing matrix analysis was used to examine the packet switched models. The results of the mathemtical analysis was discussed in Chapter II. The analysis was concerned with the average time to transmit a message/packet from node 1 to node 6 in the circuit switched model and from node 1 through node 7 in the packet switched model. The results of the mathematical model and the simulation model are shown in Table(7).

TABLE 7

MATHEMATICAL AND SIMULATION MODEL COMPARISON

Network Type (number)	Service Rate (pkt/sec)	Math Delay Time (sec)	Simulation Delay Time (sec)	Standard Deviation (sec)
1				
Circuit Switched Voice	434.78 25.77	.0547571	.05305	.000914
Packet Switched Data	333.0	.0232126	.01942	.001766
2				
Packet Switched (combined voice interactive data and bulk data)	333.0	.0260656	.02210	.004145
Packet Switched (combined prioritized voice interactive data and bulk data)	333.0	.0256263	.0311	.00641
3				
Packet Switched (combined voice and interactive data)	333.0	.0246831	.02070	.003157
Packet Switched (combined prioritized voice and interactive data)	333.0	.0245951	.02389	.005134
Packet Switched Data	333.0	.0219396	.01924	.001749

There were several differences between the two models. Differences in delay times were accounted for by the manner in which the models were initiated. When mathematical results were calculated the system already had entities in the network, however the

simulation model started at time zero with the system empty. The largest effect this had on results was from traffic intensity. As the traffic intensity increased, the time it took a message or packet to reach its destination increased. The traffic intensity increased because packets were required to wait longer at intermediate nodes. When the simulation model started at zero, the time to travel to destination was faster because the traffic was not as intense. The average delay time was affected by these faster times which occurred at the initiation of the system.

Verification of the circuit switched and packet switched model was accomplished using a significance test. The significance test uses the equation(52:52)

$$z = \frac{\bar{X} - \mu_o}{\sigma} \qquad (3.1)$$

where

$\bar{X}$ = simulation model average delay time
μ_o = mathematical model average delay time
σ = standard deviation simulation model

The significance test compared the simulation model mean to the mathematical model mean to determine if the simulation model was accurately designed. An hypothesis was assumed in order to determine the results of this test. The hypotheses were the null hypothesis ($H_o: X = \mu_o$) and the alternate hypothesis ($H_a: X \neq \mu_o$). If the null hypothesis (Ho) was rejected then the simulation model could not be used. If the alternate hypothesis (Ha) was rejected then the simulation model was a good possibility. Testing the model at the 99 percent significance level gives a probability of α equal to (1-.99).

Rejection occurs when the (z) value falls in the critical region. Extracting a value of ($z_{\alpha/2}$) from the "Student's t-Distribution table" gave the critical region to be $|z| > 2.576$. Table 8 shows the (z) values, critical region results and hypothesis results for the mathematical model and simulation model results of Table 6 using the 99 percent significance level.

TABLE 8

Z VALUES

Network (number)	(z) values	Critical Region Results	Hypothesis Results
1			
Circuit Switched Voice	1.3238	$\|z\| < 2.576$	Ho not rejected
Packet Switched Data	-2.1235	same	same
2			
Packet Switched (combined voice interactive data and bulk data)	0.9567	same	same
Packet Switched (combined prioritized voice interactive data and bulk data)	-0.5538	same	same
3			
Packet Switched (combined voice and interactive data)	1.2616	same	same
Packet Switched (combined prioritized voice and interactive data)	-0.8365	same	same
Packet Switched Data	1.5435	same	same

Since the (z) value did not fall in the critical region then the Hypothesis (Ho), that the means are equal, was not rejected. The significance tests showed the simulation models were possible representatives of the expected results found from mathematical analysis. In the case of all simulation models the results were not rejected and therefore verified. With validation and verification of the simulation models finished a network analysis was initiated. Using the verified and validated simulation models the next chapter compared the differing schemes for network implementation.

Simplified Voice Trunking Model

The Simplified Voice Trunking Model(SVTM) was developed by Rome Air Development Center(RADC) to aid research in the area of integrated communications(20:1). The simulation model was of an integrated node which transmitted voice and data over a single path. Voice traffic was loaded on to the integrated SENET trunks and the remaining capacity was available for data(20:1). Varying loads of voice and data traffic were run over the simulation model to determine average delay time, throughput, data traffic requirements and prioritization requirements for voice under overload conditions(20:1). The SVTM represents a hybrid switching node which uses digital voice prioritized over digital data packets. The data packets were transmitted over the path during inactive periods of voice communications. Validation and verification of the SVTM was accomplished during the design phase of the simulation model and therefore was not repeated in this text(20). The results of this

simulation model are discussed in the following chapter and compared to the network topologies.

IV. NETWORK ANALYSIS

This chapter describes the method for comparison of the three network topologies and the Rome Air Development Center(RADC) study. This discussion includes the actual results of the network analysis and the results of the Simplified Voice Trunking Model(SVTM) simulation of RADC with respect to delay time, cost, througnput and power. Included in the comparison results is a discussion of the effects modifications have on the network performance and control. The comparison of the network topologies is accomplished first then these results were compared to the RADC study.

Comparison Techniques

Network Topologies. In order to perform a comparison of different network topologies it was necessary to maintain consistent conditions between each network. It was also important to keep some conditions constant within each model in order to see the effects of other conditions on performance measures. Specific conditions which were considered in the comparisons were, traffic intensity, service rates, arrivals rates, percentages of voice and data, percentages of bulk and interactive data, priorities, milage costs, voice digitization costs and switching costs. Traffic intensities and arrival rates were maintained as constants throughout the analysis. Traffic intensity was kept constant in order to test the effects of set conditions on a fully loaded communications system. Controlling

the flow of entities allowed the traffic intensity to be kept constant throughout the system, as explained in chapter II page(40). Arrival rates were maintained the same in all situations because the same basic effect arrival rates had on the system was observed by varying the service rates. Arrival rates were derived as shown on page (35) of chapter II. These arrival rates were based on packet size, channel capacity, header size and percentage of bulk data over interactive data. Arrival rates for each network are shown in Table 9.

TABLE 9

ARRIVAL RATES

Network Type	Arrival Rate (pkt/sec)
1	
Circuit Switched Voice	20
Packet Switched Data	20
2	
Packet Switched Voice	20
Interactive Data	11.04
and Bulk Data	8.96
3	
Packet Switched Voice	20
and Interactive Data	11.04
Packet Switched Bulk Data	8.96

The next step in the comparison analysis was to establish the procedures for evaluating the networks on varying conditions.

The method used to evaluate the other conditions was by using a step by step process. The first step necessary for analyzing the

variable conditions was to evaluate the effects of a varying service rates. The initial service rate was 333 packets per second for the packet switched network the average node service rate of the DDN. The service rate for the circuit switched model was .0023 seconds per packet for the reservation signal and .0333 seconds per packet for message transmission. The lowest service rate was not allowed to go below the service rate which permitted rejections of one percent. The lowest allowable service rate was then maintained as a constant to evaluate the effects of voice and data percentages and bulk versus interactive percentages. The next step in this evaluation was to prioritize voice over data and bulk data over interactive data. The key percentage combinations are show in Table 10 based on networks 2 and 3.

TABLE 10

PERCENTAGES COMBINATIONS

Network	%Voice/Data	%Bulk/Interactive
3 Packet Switched (combined voice and interactive data)	75/25	
	90/10	
	10/90	
2 Packet Switched (combined voice interactive data and bulk data)	75/25	
	75/25	90/10, 25/75, 10/90
	90/10	
	90/10	90/10, 25/75, 10/90
	10/90	
	10/90	90/10, 25,75, 10/90

In each step of these procedures it was important to evaluate the effects the changed conditions have on the network performance.

The final comparison of the networks was a cost analysis of mileage, voice digitization and switching nodes. Milage costs were based on specific distances between nodes, voice digitization costs were based on the channel capacity used by digital voice, and switching costs were based on type and quantity of nodes. These costs were calculated for a path length of 6 primary nodes. Path lengths can be altered by changing source or destination attributes. Cost results were straight forward and used only as a general cost ratio comparison rather than actual network costs. The overall performance criteria to evaluate in each level of comparison was delay time, cost, throughput and power. Once the inputs and the method of comparison was established, the simulations were run and results analyzed.

Simplified Voice Trunking Model. Performing a comparison of the network topologies using SLAM with the integrated switching node of the SVTM was necessary to evaluate the capabilities of hybrid switching techniques. Iiyas(40) provided a means for comparing these two simulation models. This means of comparison was found in the managers tool for performance measurement, power(40:19). Using the throughput and average delay time from the SVTM simulation and the network topologies a comparison was accomplished using the performance criterion power in Chapter II.

Network Topology Results

The results of the simulation analysis was based on specific models, as well as specific networks. The three networks were broken down into five models. These models were analyzed separately and then together as networks. The first network consisted of a circuit switched model and a single packet switched model. The second network was a single multiple arrival packet switched model. The third network was a dual arrival packet switched model and a single arrival packet switched model.

The first step in the collection and analysis of results was accomplished by varying the service rates of the five models. The initial service rate for all packet switched models was 333 packets per second. This rate was inverted for use in the simulation models to .003 seconds per packet. The circuit switched model used two service rates which were determined using the message size, header size and channel capacity. The initial rate for the circuit switched model was .0023 seconds per packet for the securing of the path and .0333 seconds per packet to request transmit, transmit the message,and release the path. The results of three service rates per model are shown on Table 11.

TABLE 11

NETWORK COMPARISON I

Network Type (number)	Service Rate (pkt/sec)	Packet Loss (pkt)	Delay Time (sec)	Through-put (pkt/sec)	Power (pkt/sec-sec)
1 Circuit Switched Voice	25.77	0	.05309	1.315	24.77
	333.0	0	.01721	1.6	92.97
Packet Switched Data	1000.0	0	.00630	1.79	284.13
	333.0	0	.01942	1.72	88.56
	100.0	3	.07461	1.69	22.65
Total(1)	333.0	0	.01942	3.22	165.81
2 Packet Switched (combined voice interactive data and bulk data)	500.0	0	.01343	3.32	246.29
	333.0	0	.02210	3.46	156.33
	250.0	2	.03315	3.42	103.17
3 Packet Switched (combined voice and interactive data)	1000.0	0	.01293	2.55	196.83
	333.0	0	.02070	2.76	133.33
	200.0	3	.04069	2.64	64.38
Packet Switched Bulk Data	1000.0	0	.01271	.795	62.55
	333.0	0	.01924	.745	38.72
Total(3)	333.0	0	.02070	3.51	159.32

Analyzing the models shown in Table 11 showed that the service rate had a great impact on delay time, throughput and packet loss. Decreasing the service rate slows down the packets/messages traveling through the network. When the packets/messages slowed down the delay time increased, the throughput decreased and the queues increased

causing more packet rejections. Decreased service rate greatly increased delay time, depending on traffic intensity in a network(see Figure 7).

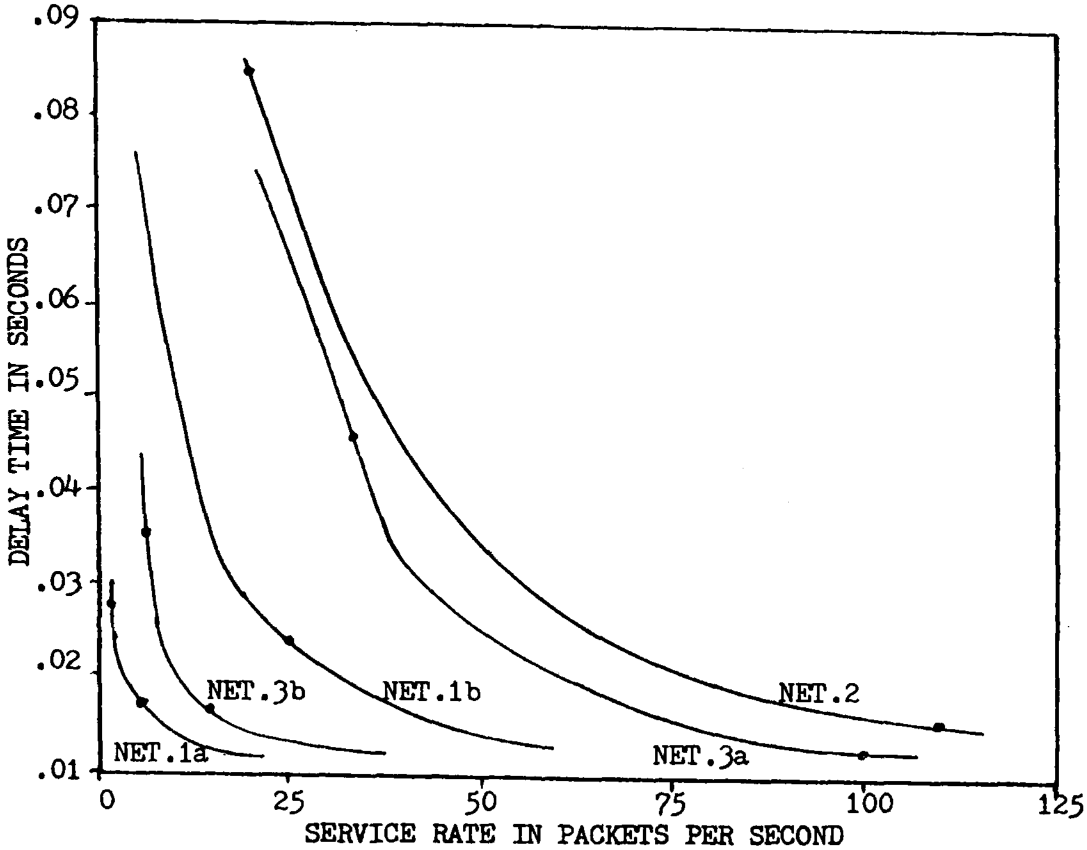

Figure 7. Service Rate versus Delay Time for Network Topologies
Networks maintain constant ρ and arrival rates.

The decreased service rates affected the network performance of network 2 and network 3a first. This decreased network performance depended on the traffic intensity(saturation level) of the network. Network 2, a network transmitting digitized voice, interactive data and bulk data over a single packet switched path, was the network

with the highest saturation level. This network had more arrivals than other models. Network 3a, the model which transmitted digitized voice and interactive data for the third network, had the next highest saturation level. Because of the high saturation levels network 2's and network 3a's delay time increased faster than the two bulk data models, network 1b and 3a.

The service rate affected throughput also(see Figure 8).

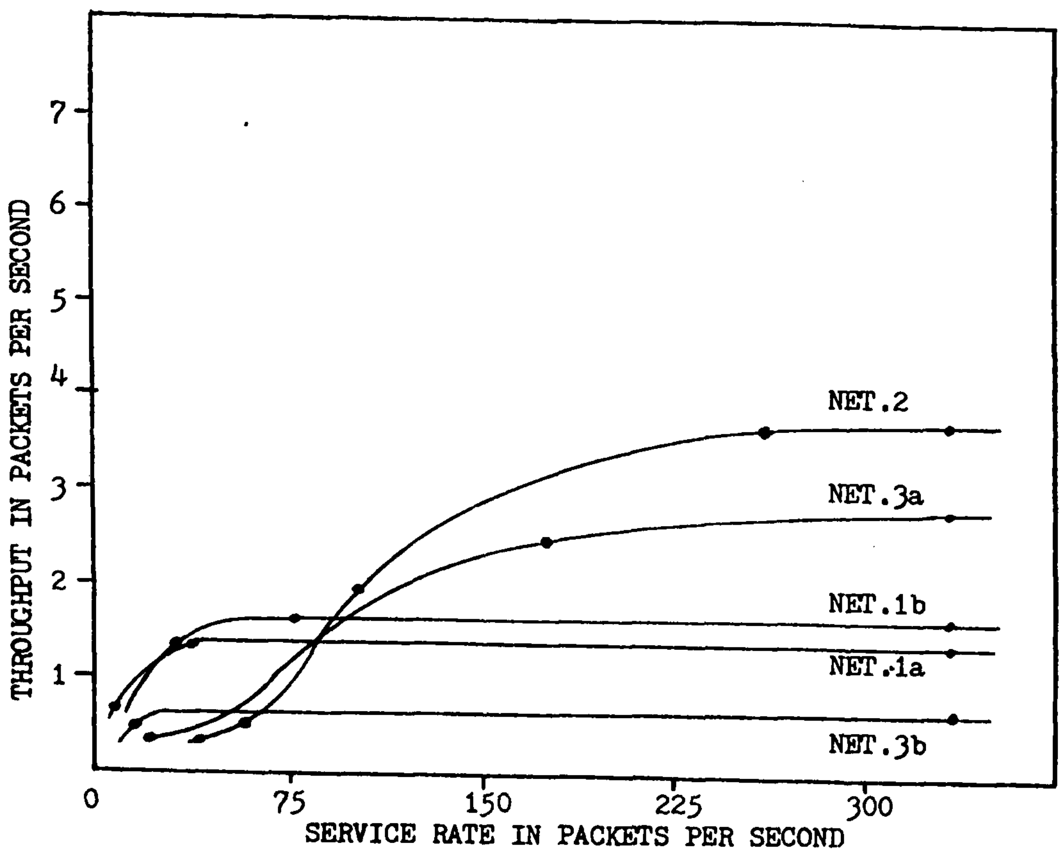

Figure 8. Service Rate versus Throughput for Network Topologies Networks maintain constant ρ and arrival rates.

As service rates increased throughput decreased, because fewer packets were allowed through the system during a given time period.

The networks with the highest saturation levels, network 2 and network 3a, had the highest throughput, because they had better utilization of resources. Since their saturation levels are higher they decreased in throughput at a faster rate than the lower saturated network models. The lower saturated models, network 1b and network 3b, had a slower decrease in throughput because they had a much lower throughput to begin with, as compared to networks 2 and 3a. Since the circuit switched network reserved the entire network to transmit one message it depended on fast service rates to maintain a reasonable throughput performance. When service rate for the message transmission increased overall system performance was decreased.

The circuit switched network takes a much longer time to reach the first rejection due to the dual service rate configuration. The small header was serviced easily without causing rejections and the message transmission which used the larger service rate is only processed once. Although packet loss was difficult to achieve it became unimportant because of the rapid decrease in throughput shown in Figure 8. The packet switched networks, however, achieved packet loss more rapidly when the service rate decreased(see Figure 9).

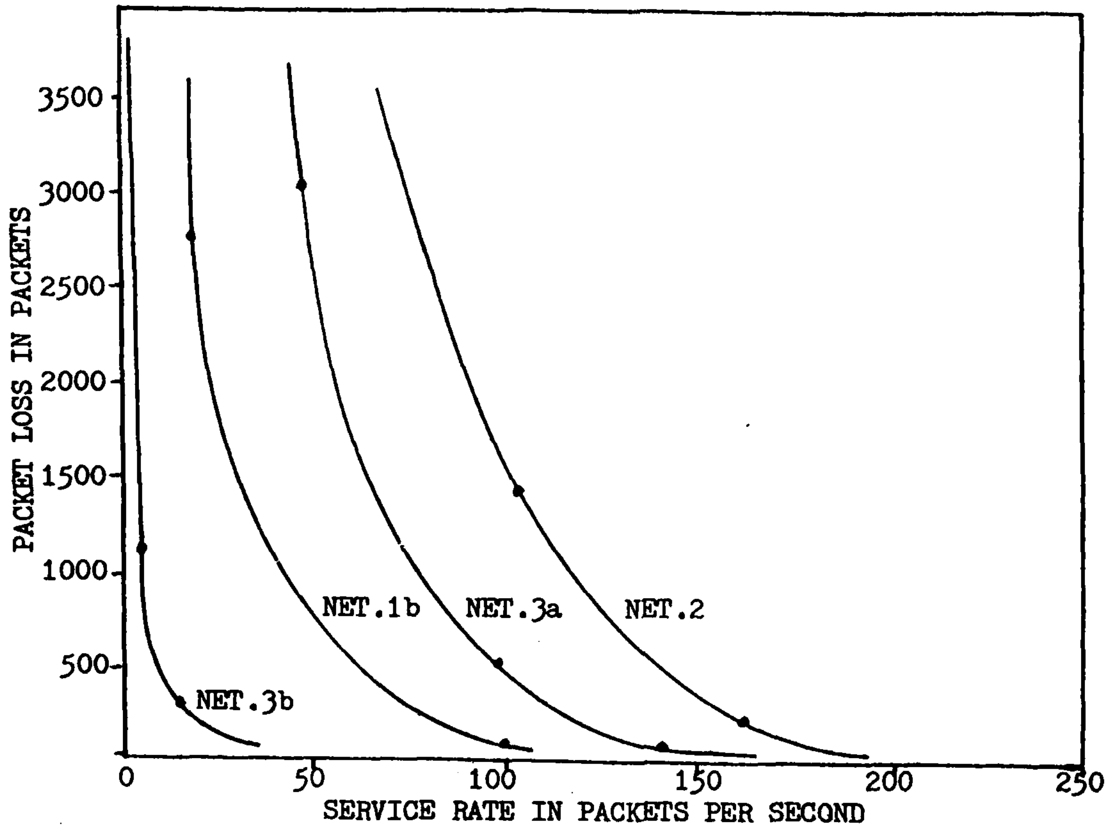

Figure 9. Service Rate versus Packet Loss for Network Topologies Networks maintain constant ρ and arrival rates.

As seen in Figure 9 the packet loss increased at a faster rate for the more saturated packet switched networks of 2, the network transmitting digitized voice, interactive data and bulk data over a single packet switched path, and 3a, the network transmitting digitized voice and interactive data over a single path packet switched path. The two bulk data packet switched paths, network 1b and network 3b, did not achieve packet loss as quickly as the other packet switched networks. Since these two networks are less

saturated, due to the lower arrival rates, the number of rejections did not increase as rapidly as the other two networks, when the service rate increased. With or without a high saturation level all networks became ineffective quickly if analyzed from perspective of the maximum number of rejections for the network to efficiently function. Based on the one percent rejection rate allowed for an efficient system, the service rate must be kept high.

Combining the models and analyzing the three networks shows significant results in the area of throughput. The delay time differences were minimal between the three networks. In the case of two separate paths the greater delay time was used as the total time and the number of packets summed for the total throughput. The second and third network provided higher throughput, however over a longer period of time the third network would provide the higher output. Analyzing the arrival rates from Table 9 shows that the second network delay time though slightly lower provided the better utilization of network resources and was the better choice in this section of the analysis.

The next step in the comparison results was to evaluate the effects of percentages on the various models. The lowest allowable service rate was used in order to get the effect of percentage changes on packet rejections. The portions of networks not affected by the percentages of voice and data were still included in the total network summation. Table 12 gives the results for the percentage analysis.

TABLE 12

NETWORK COMPARISON USING PERCENTAGES

Packet Switched Digital Voice, Interactive Data and Bulk Data.

Network Type (number)	Service Rate (pkt/sec)	%Voice to Data	%Bulk to Inter.	Packet Loss (pkt)	Delay Time (sec)	Through-put (pkt)	Power (pkt/sec-sec)
(2)	333.0	75/25	-	0	.02132	3.3	151.24
	250.0	75/25	-	3	.03272	3.35	102.23
	1000.0	75/25	90/10	0	.01349	3.21	237.95
	1000.0	75/25	25/75	0	.01355	3.41	251.66
	333.0	75/25	25/75	1800	.02156	3.22	143.43
	1000.0	75/25	10/90	0	.01384	3.25	234.33
	333.0	90/10	-	0	.02131	3.43	157.27
	250.0	90/10	-	2	.03294	3.33	101.09
	1000.0	90/10	90/10	0	.01350	3.21	237.77
	1000.0	90/10	25/75	0	.01351	3.42	252.77
	1000.0	90/10	10/90	0	.01354	3.23	238.18
	333.0	10/90	-	0	.02174	3.43	157.54
	250.0	10/90	-	2	.03312	3.37	101.6
	1000.0	10/90	90/10	0	.01351	3.21	237.6
	1000.0	10/90	25/75	0	.01351	3.29	243.52
	1000.0	10/90	10/90	0	.01357	3.4	250.55

Packet Switched Digital Voice and Interactive Data.

Network Type (number)	Service Rate (pkt/sec)	%Voice to Data	%Bulk to Inter.	Packet Loss (pkt)	Delay Time (sec)	Through-put (pkt)	Power (pkt/sec-sec)
(3a)	333.0	75/25	-	0	.02082	2.77	132.81
	200.0	75/25	-	5	.04067	2.52	61.96
	250.0	90/10	-	2	.02948	2.55	85.5
	333.0	10/90	-	0	.02090	2.59	124.16
	250.0	10/90	-	2	.02967	2.6	87.63

Altering the percentages of voice and data entering the network only affected networks 2 and 3. Changing the percentages of bulk data versus interactive data only affected network 2. The comparison of network 2 and the various combinations of voice to data and bulk to interactive data show very little change in the average delay

times but a marked difference in throughput results, because the arrival rate of bulk and interactive data are similar. As voice percentage decreased over data percentages the throughput increased with the lower bulk. With the increase of voice over data the throughput peaked with a lower percent bulk and then rapidly decreased. Assuming voice to be the higher percentage of arrivals would reccommend a medium to lower range for percent bulk data to maximize network 2 throughput.

Analyzing the overall networks 2 and 3 using percentage shows that in the extremes of percent voice over data and percent data over voice there was a sufficient increase in throughput for the second network. However in the medium ranges of percentages the third network surpassed the second network in average delay time and throughput. Assuming medium range percentages of voice and data,for example 75/25, the third network was the better choice of voice and data communications.

Prioritization was the next step in gathering and analyzing results. The methods used for prioritization in SLAM was low value first (LVF) and high value first (HVF) The value examined with this scheme was the meaninterarrival rates. In the network which used a single path for digitized voice and data the given arrival rate of voice at .05 seconds per packet was the low value and the arrival rate of bulk data at .112 seconds per packet was the high value. Since interactive data had an arrival rate of .091 seconds per packet it falls between the two in terms of priority. The third network which integrated voice and interactive data had voice as the low value and interactive data as the high value. Simulation models for

these networks were analyzed to evaluate the effect priorities have on other parameters of the network. Using the scheme (LVF) and (HVF) provided priority to voice if using (LVF) and data if using (HVF). These two networks were analyzed using these priority schemes with two different methods, interrupt and noninterrupt. The interrupt scheme prioritizes one packet over the other packet even if the other was being processed, causing the inprocess packet to be preempted and wait to repeat processing. The noninterrupt prioritizes one type packet over the other, without interruption of a packet already being processed. The noninterrupt scheme was accomplished using the secondary priority scheme discussed on page (57) of chapter III. The results of this evaluation are shown in TABLE 13.

TABLE 13

NETWORK COMPARISON USING PRIORITIES

Packet Switched Digital Voice, Interactive Data and Bulk Data.

Network Type (number)	Service Rate (pkt/sec)	Interrupt (yes/no)	Priority Type (LVF/HVF)	Packet Loss (pkt)	Delay Time (sec)	Through-put (pkt/sec)
2	250	no	LVF	2	.03106	106.41
2	"	yes	LVF	0	.03111	103.85
2	"	no	HVF	2	.03435	97.96
2	"	yes	HVF	0	.03469	89.94

Packet Switched Digital Voice and Interactive Data.

Network Type (number)	Service Rate (pkt/sec)	Interrupt (yes/no)	Priority Type (LVF/HVF)	Packet Loss (pkt)	Delay Time (sec)	Through-put (pkt/sec)
3	200.0	no	LVF	4	.03837	65.22
3	"	yes	LVF	2	.03863	64.46
3	"	no	HVF	4	.04235	61.75
3	"	yes	HVF	2	.04222	57.2

Comparing Table 11 results to Table 13 show some significant changes to delay, throughput and speech quality. Prioritizing voice over data results in a decrease in throughput, however since delay also decreases this was an improvement in performance as noted by the power criterion. When using the interrupt scheme this improvement was seen in another area. Power performance remained the same, there was a decrease in packet loss. When prioritizing data over voice the overall performance was decreased in both cases of interrupt scheme. The only advantage seen by prioritizing data over voice was in the interrupt scheme which showed a decrease in packet loss, however an even greater decrease was seen in overall performance as noted by the power criterion. Prioritizing voice over data showed an improvement to system performance because there were more voice packets entering the system. Prioritizing data over voice was in effect prioritizing the smaller number of packets in the system over the larger number of packets, therefore causing a detrimental effect on system performance.

The results of the simulation model show that each type of network has its particular advantages and disadvantages. Prior to addition of percentages and prioritization networks 2 and 3 were in a close running for the better network performance depending on the users view of the poor utilization of the third network. Once percentages were implemented into the system network 3 provided the better performance in the medium ranges of voice and data percentages. Prioritization ,on the other hand, shows network 2 as the superior model especially in terms of the power criterion. The final outcome depends on the specific user requirements and desired

results for network performance.

Simplified Voice Trunking Model Results

The Simplfied Voice Trunking Model integrated switching node based results on variations of traffic loads of voice and data. Several scenarios were run which varied the number of voice transmissions and the arrival rates of data packets. The SVTM simulations were analyzed by comparison of the number of voice calls, data arrival rates, average delay time and throughput. The next step in the analysis was to calculate networks results using the integrated node switching results of the SVTM. The network results consisted of a six node network.

The first step in the analysis was a comparison of six different scenarios as shown in Table 14.

TABLE 14

SIMPLIFIED VOICE TRUNKING MODEL RESULTS

SVTM sim. (#)	Average Arrival Rate (pkt/sec)	Average Delay Time (sec)	Throughput Voice (pkt/sec)	Throughput Data (pkt/sec)
1	.340	.0121	.0592	2.45
2	1.395	.0147	.108	2.41
3	.643	.0131	.099	1.57
4	.163	.0115	.09	1.31
5	.223	.0123	.108	.81
6	.570	.0126	.092	1.93

As the arrival rate decreased so did the number of voice calls(packets) and/or the number of data packets. An analysis of the

results of Table 14 was accomplished by first comparing two SVTM results with constant voice throughput, then comparing two SVTM results with constant data throughput. This was shown in Figures 10 through 13. The first comparison was an analysis of data throughput with varying arrival rates and constant voice throughput. Maintaining a constant voice throughput of .108 and varying arrival rates was represented by a graph of voice calls to time in minutes(see Figure 10).

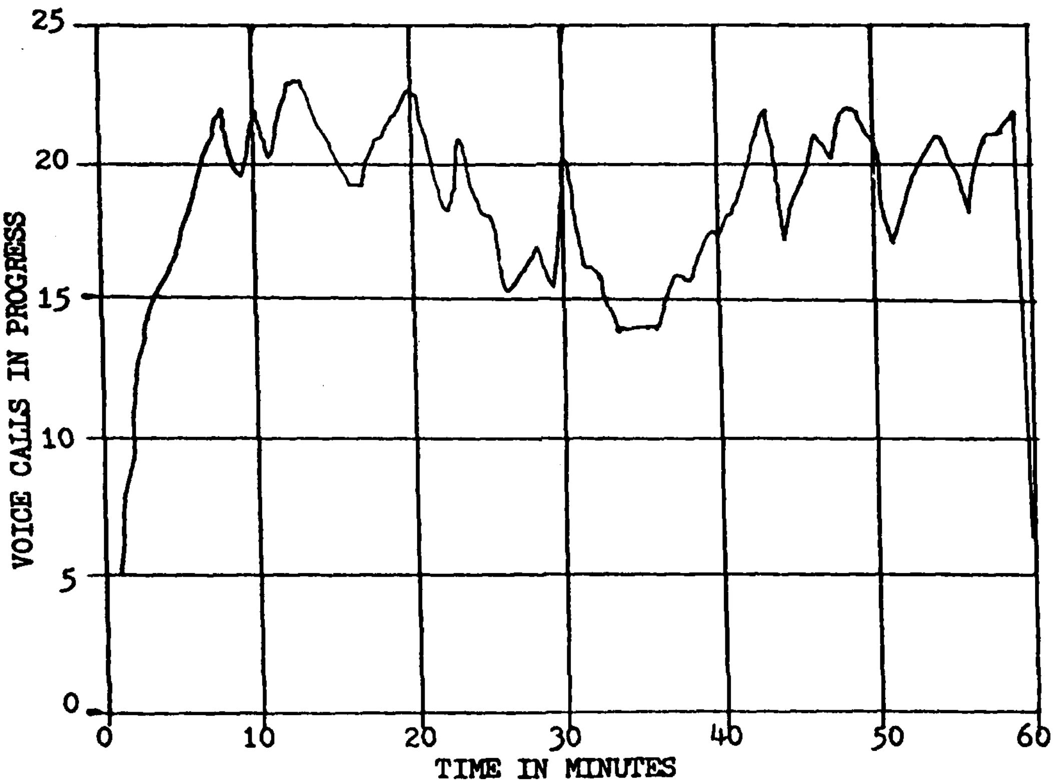

Figure 10. Ratio of Voice Throughput(.108) to Time

The constant voice throughput with varying arrival rates effects the

data throughput. When arrival rates were high the network maintained a high throughput(see Figure 11).

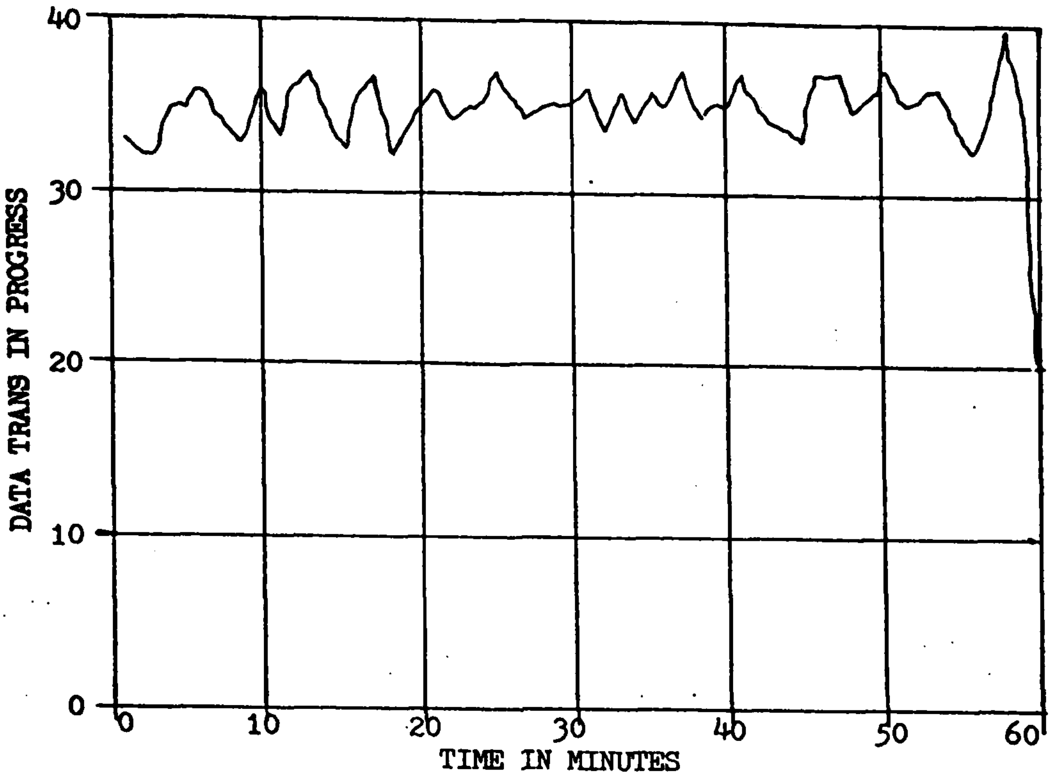

Figure 11. Data Throughput (2.41) with Voice Throughput(.108)

The second comparison was acomplished by keeping voice calls constant. Using decreased arrival rates had a detrimental effect on data throughput(see Figure 12).

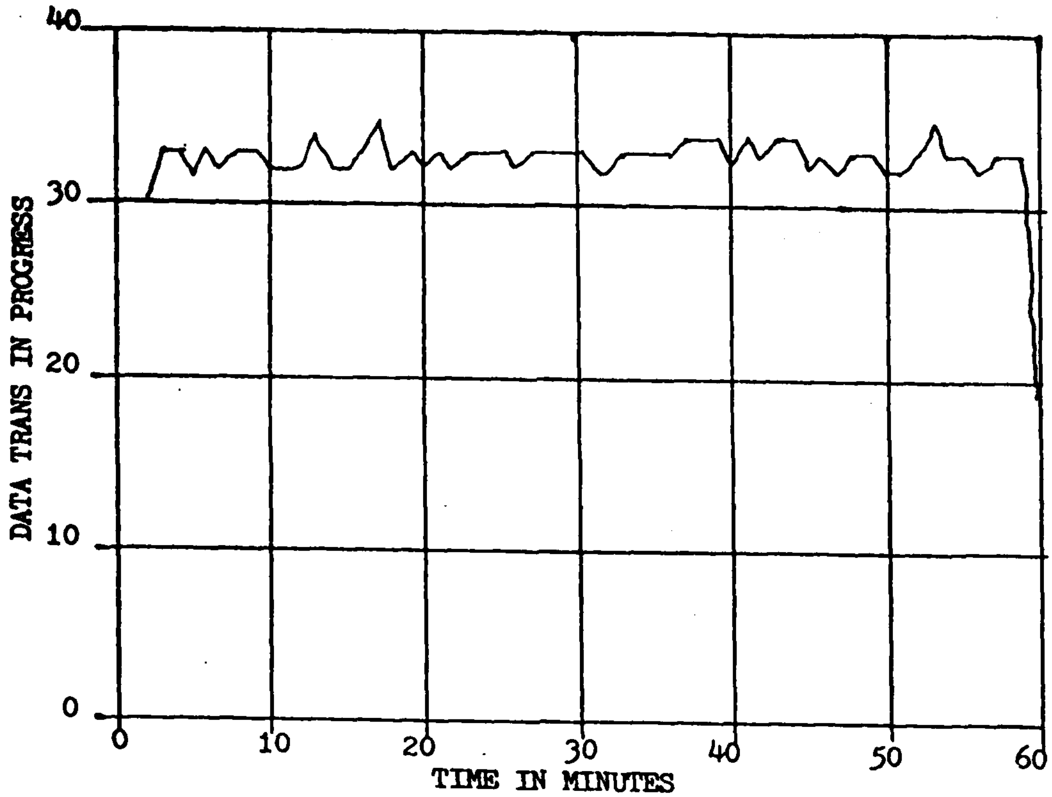

Figure 12. Data Throughput(0.81) with Voice Throughput(.108)

Figure 11 and 12 showed the effect higher arrival rates of data had on data throughput. With a set voice rate the more efficient performance (higher throughput) was seen using the higher arrival rate. The lower arival rate as shown in Figure 12 shown poor utilization of the channel.

Analyzing the situation in which the data packet throughput remained constant showed that a decrease in the required number of voice calls allowed the system to become more efficient. Maintaining data throughput constant and using an arrival rate of .340 packets per second showed identical results to the data throughput of Figure

11 which used and arrival rate of 1.896 packets per second(see Figure 13).

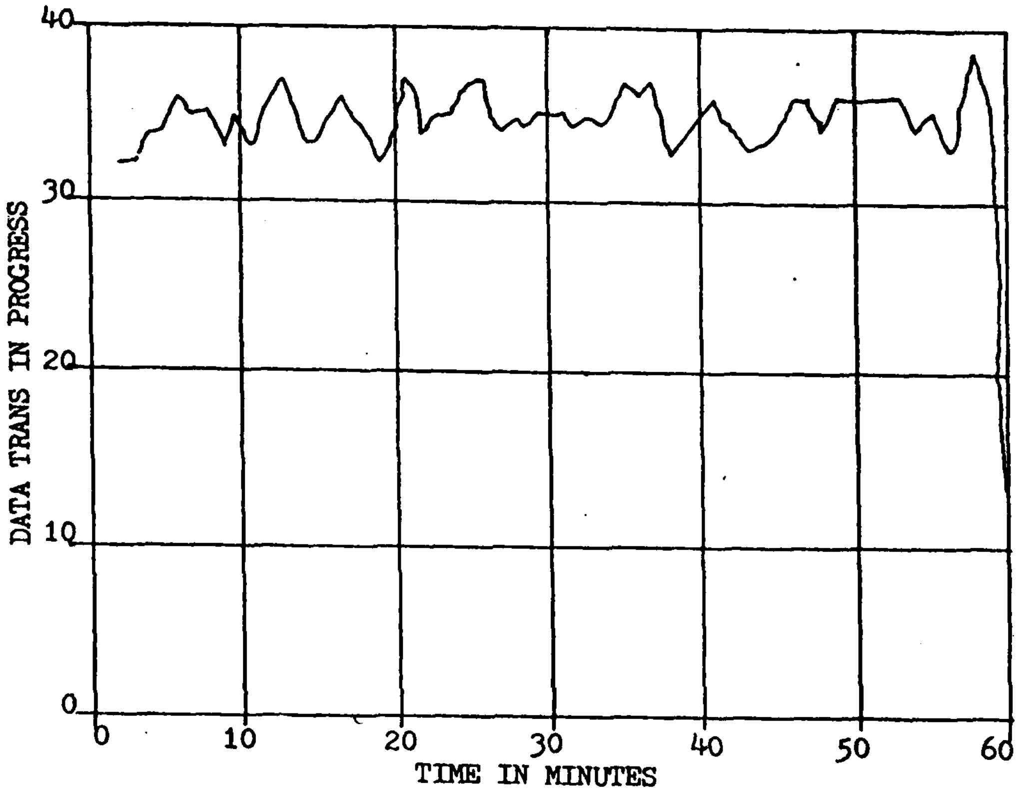

Figure 13. Data Throughput(2.45) with Voice Throughput(.0592)

The results of maintaining high constant data throughput with varying arrival rates showed the effect the network had on voice throughput(see Figure 14).

AD-A164 176 ANALYSIS OF INTEGRATED AND NONINTEGRATED VOICE AND DATA NETWORKS FOR DOD C (U) AIR FORCE INST OF TECH WRIGHT-PATTERSON AFB OH SCHOOL OF ENGI.. S L WALKER 2/2

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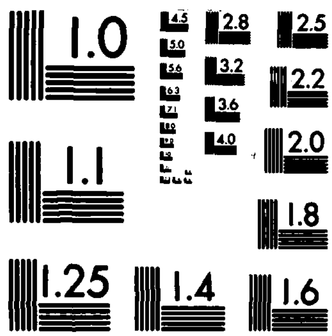


MICROCOPY RESOLUTION TEST CHART

ƆARDS-1963-A

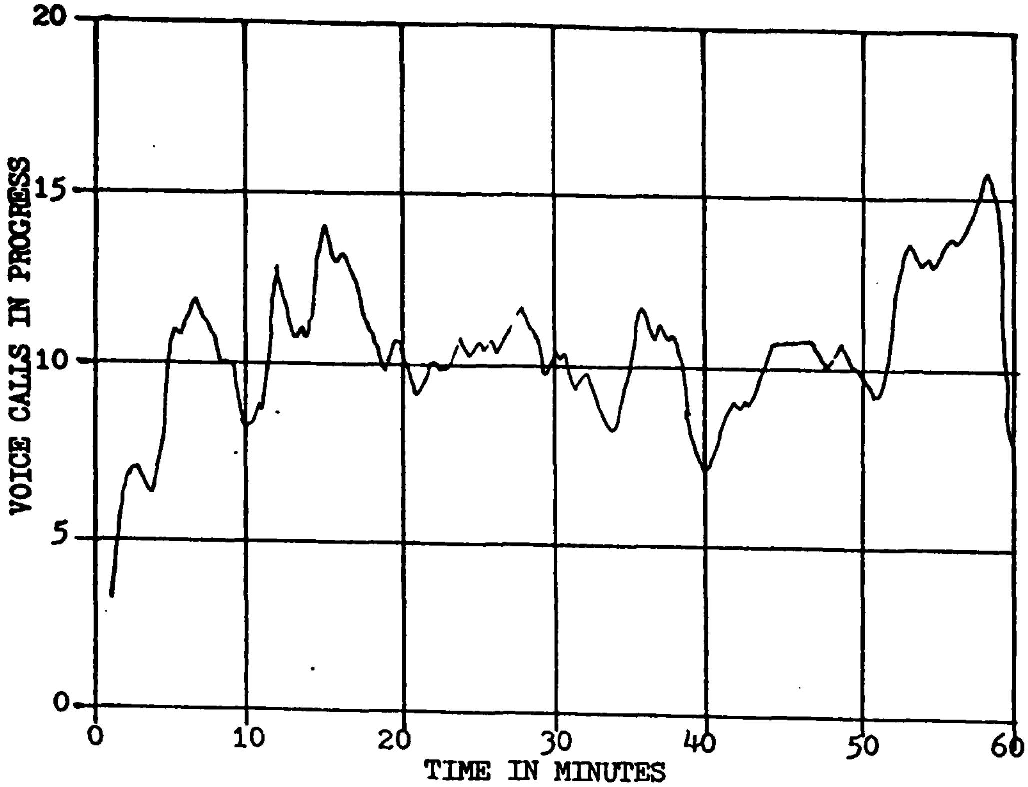


Figure 14. Ratio of Voice Throughput(.0592) to Time

Comparing Figure 14 to Figure 10 shows a drastic decrease in voice throughput, however both systems were similar in overall throughput because of the varying arrival rates. A comparison of the first scenario, maintaining voice throughput constant, with the second scenario, maintaining data throughput constant, showed the second had the more efficient performance when varying the data arrival rates.

The final step in the SVTM analysis was to calculate the average delay time for a network containing six nodes. This network calculation provided the necessary information for a comparison with

network topologies. The throughput of the network was the same as the single node calculations. The SVTM network calculations are shown in Table 15.

TABLE 15

SVTM NETWORK RESULTS

SVTM sim. (#)	Average Delay Time (sec)	Total Throughput (pkt/sec)	Total Power (pkt/sec-sec)
1	.0726	5.00	69.0
2	.0882	5.01	56.8
3	.0786	1.67	21.2
4	.069	1.40	20.3
5	.0768	0.92	12.0
6	.0756	2.02	26.7

This analysis shows the higher performance with scenario 1. From the previous analysis scenario 1 had a much lower arrival rate and number of voice calls, however analyzing the power criterion shows scenario has the better performance results. Scenario 1 which produced the higher performance results was used in the comparison with the circuit switched voice/ packet switched data network and the packet switched digital voice and data networks. The power performance measurement was the criterion used for comparing the network topologies with the hybrid switched simulation model from RADC.

Cost Results

Cost analysis was discussed in Chapter II. The mileage, switching and VDR cost were basic calculations which depend on

mileage, type of node, number of nodes and whether digitized voice was required. Additional costs such as those acquired from increased service rates was not be addressed. Costs acquired through buffer space was not addressed since all buffers were maintained at a set capacity to verify rejection rate. The cost analysis of chapter II was separated into three networks and shown in Table 16.

TABLE 16

NETWORK COST ANALYSIS

Network Type (number)	Mileage (millions of dollars)	Voice Digit. Rate(millions of dollars)	Switching (millions of dollars)	Total Cost (millions of dollars)
1 Circuit Switched Voice/ Packet Switched Data	.2416	2.34	1.26	3.842
2 Packet Switched (combined Voice Interactive Data and Bulk Data)	.1208	2.64	.420	3.181
3 Packet Switched (combined Voice and Interactive Data)/ Packet Switched Bulk Data	.2416	2.88	.340	3.962
Simplified Voice Trunking Model	.1208	2.34	.238	2.699

In cost analysis it was important to note that these figures only gave a general idea of a ratio comparison of the three networks. The first network cost was larger than those of the other networks because of the increased switching costs. The third network exceeded

the second network because of the additional milage and switching nodes of the second path.

Conclusion

The results of the three network topology analysis was compared to the results of the Simplified Voice Trunking Model analysis. Using average delay time and throughput the performance criterion power was determined and displayed in Table 17.

TABLE 17

POWER COMPARISON

*All networks have a ρ equal to .75.
*Total arrival rates for networks 1 through 3 equal 20 pkt/sec and
*the arrival rate for the SVTM is .34 pkt/sec.

Network Type (number)	Average Delay Time(sec)	Through-put (pkt/sec)	Power (pkt/sec-sec)	Cost (millions of dollars)
1 Circuit Switched Voice Packet Switched Data	.01942	3.22	165.81	3.842
2 Packet Switched Voice Interactive and Bulk Data	.02210	3.46	156.33	3.181
3 Packet Switched Voice and Interactive Data Packet Switched Bulk Data	.02070	3.51	169.32	3.962
Simplified Voice Trunking Model	.07260	5.00	69.0	2.699

Table 17 shows the results of the network analysis and the solution

to the hybrid switched Simplified Voice Trunking Model(SVTM). The SVTM results show a network with a power rating of 69.0 packets per second-second.

The three networks analyzed in this research use similar arrival rates to that of the higher SVTM scenario as compared to the other scenarios. Network 3, a digital voice and interactive data packet switched network with a separate packet switched path for bulk data, had the highest power rating other than the higher SVTM. The third network exceeded the first network in performance because of the higher throughput which occurred through the two packet switched paths versus a packet switched path and a circuit switched path. The third network surpassed the second network, voice and data integrated over a single path, due to the lower utilization of the path. The third network was not as saturated, therefore the delay time for packets was lower. The disadvantage with network 3 was its high cost and inefficient utilization on the bulk data packet switched path. The next highest power rating is the first network, a network using separate digital voice and digital data paths. The disadvantage of this network was the inefficient use of the circuit switched path as discussed in the first chapter and the highest cost of all three networks. The second network, a single path integrated digital voice and data network, had the lowest power rating but was the most efficient of the three networks. The second network did not have the poor utilization caused by the circuit switched voice network or waste of resources as in the third network caused by the bulk data path.

Of the three networks the second network was the best overall

choice of the network topologies because of the lower costs and efficient use of network resources, however it does not allow for future growth without increasing capabilities. Overall the hybrid switched SVTM with circuit switched digital voice using the same channel as packet switched data provided the better cost results. The power measurement of 69.0 packets per second-second was the lowest power rating because of the large delay time. The large delay time is attributed to the time data packets must wait for the transmission of voice. The use of a network designed after the Simplified Voice Trunking Model would be beneficial to an environment with a decreased number of voice calls and high data input.

V. CONCLUSIONS AND RECOMMENDATIONS

The final chapter of this research discusses the results, conclusions and recommendations. This research was accomplished in order to determine the best switching approach with or without voice and data integration would best meet DOD communications requirements.

Selecting the appropriate switching techniques for DOD communications was very difficult if not impossible to accomplish. Due to varied requirements and needs it is possible there is no one means of switching technique which is best for DOD communications. The difficulty in selecting a switching approach was in pinpointing specific communications requirements. The scope of the problem was narrowed to evaluating specific input data presently used over the Defense Data Network (DDN) to determine the better switching approach for voice and data communications.

Conclusion

The possible approaches to use in finding a solution to this problem was circuit, packet or hybrid switching techniques over an integrated or nonintegrated communications path. Three network topologies and the Simplified Voice Trunking Model(SVTM) from Rome Development Center were analyzed in this research. The first network topology had separate path for digital voice and digital data. Digital voice was transmitted over a circuit switched network and digital data was transmitted over a packet switched network. The

second network consisted of a single packet switched path for transmitting digital voice, interactive data and bulk data. The third network topology also used two separate paths. The first path was a packet switched path to transmit digital voice and digital interactive data. The second path was a packet switched path for transmitting digital bulk data. The SVTM simulation used a single path for transmission of digital voice and digital data.

The first point of consideration was whether to use voice and data integration or to use separate paths. Based on research and growing technology voice and data integration is the direction for future communications. From this research, integrated voice and data networks using the packet switching technique showed better performance results than separate voice and data communications networks or hybrid switching technique. This agrees with the research accomplished by Gitman and Frank(30) which found voice and data integration was the best approach to take for DOD communications. This research confirms the need for DOD to direct efforts towards integrated networks because of the more efficient handling of voice and data.

The second point of consideration concerns the type of switching technique best for the DOD switching techniques. This research agrees in part with Gitman and Franks(30) final conclusion that packet switching is a better approach for DOD communications. From this research the packet switching technique with a single path for digital voice and digital data provides the highest power value with lower cost than the separate paths for voice and data or the hybrid switching network. Significant to this conclusion is the necessity

of creating an integrated voice and data packet switched network capable of allowing for future growth. This is important because the network which uses separate voice and data paths allow for future growth but the network which uses a single packet switched path for voice and data is already saturated when using the same input parameters. The hybrid switching network represented by the SVTM showed the lowest power rating but this was mainly due to the lower average arrival rate. The hybrid switching network could be effective especially for those networks requiring higher percentages of data transmissions and priority voice transmissions.

Because of the potential of the network using a single packet switched path for voice and data and the network using a hybird switched path for voice and data the consideration of an integrated voice and data network should not be overlooked. Although this research did not show the capability for future growth in the voice and data integrated networks, their lower cost and higher throughput have a high potential for meeting the future needs of increasing communications, lower costs and better use of resources.

Recommendation

The recommendations from this research are for continued analysis using the simulation models and creation of a simulation model network using the SVTM simulation results. The recommendations include analysis of all performance variables as well as evaluation of different assumptions.

It is recommended that continued analysis of the simulation

models include evaluation of performance variables maintained constant in this research. Specific parameters which should be altered and evaluated include packet size, channel capacity, number of servers, arrival rates, path algorithm, and utilization. Packet size should be varied to determined the packet size and arrival rate which maximizes the system. Altering the channel capacity and number of servers provides information on transmitting packets simultaneously over a channel. Altering the path algorithm requires evaluating the percentage of shortest path algorithm and making modifications to the simulation model to create variable alternate paths. This modification requires a much larger model with additional nodes. The other limitation of this modification is the increased number of entities in the simulation system required by the additional number of nodes.

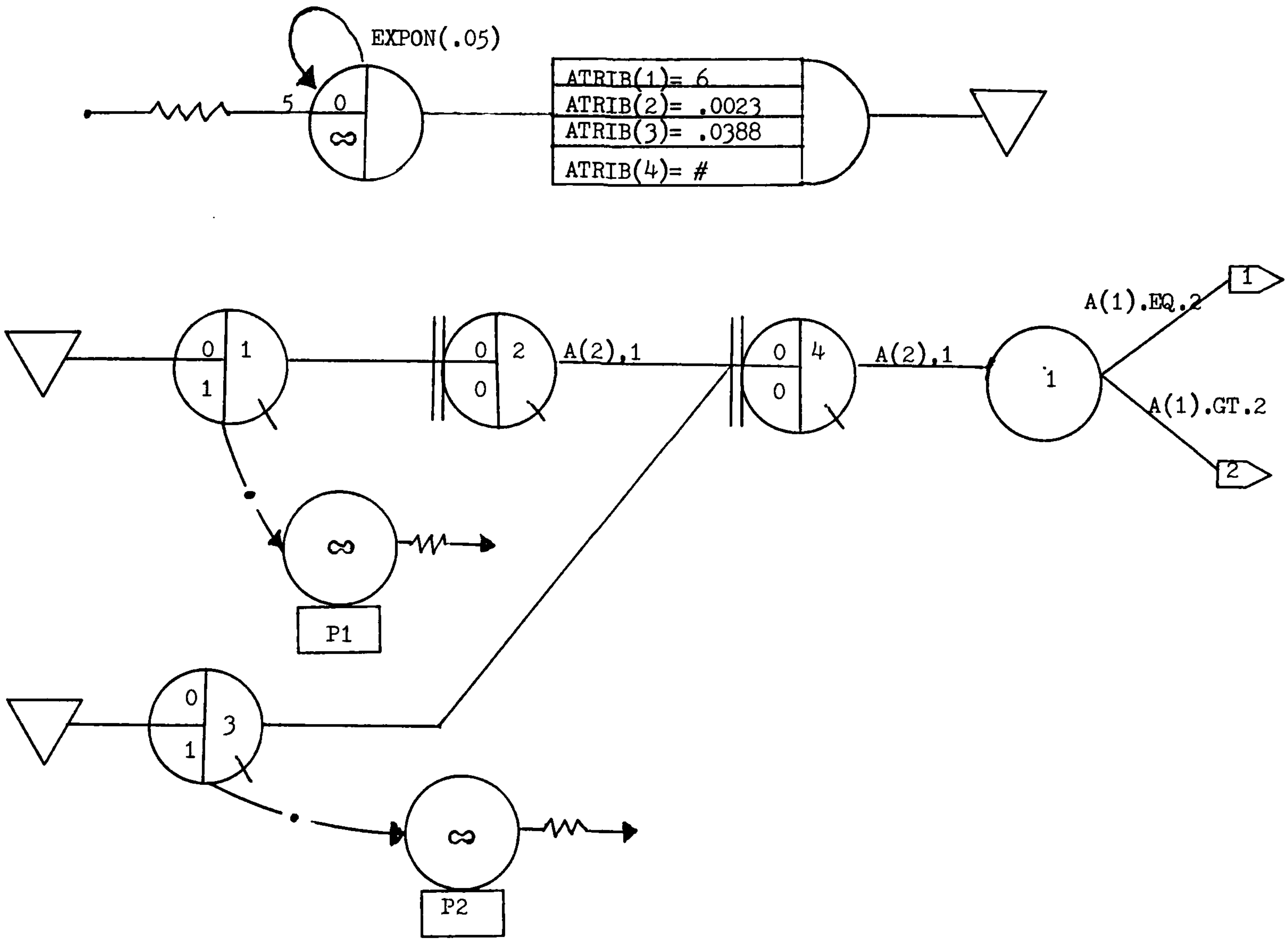
The other recommendation is to create a network model to represent the results of the SVTM node simulation. A network with exact replicat nodes as used in the SVTM would be extremely time consuming in evaluating results, because of the extensive time required to run the single node evaluation. However, a network simulation using a limited version of the SVTM node would be beneficial to an overall performance evaluation of the hybrid switching network. This analysis should include the range of performance parameters discussed in this research.

APPENDIX A

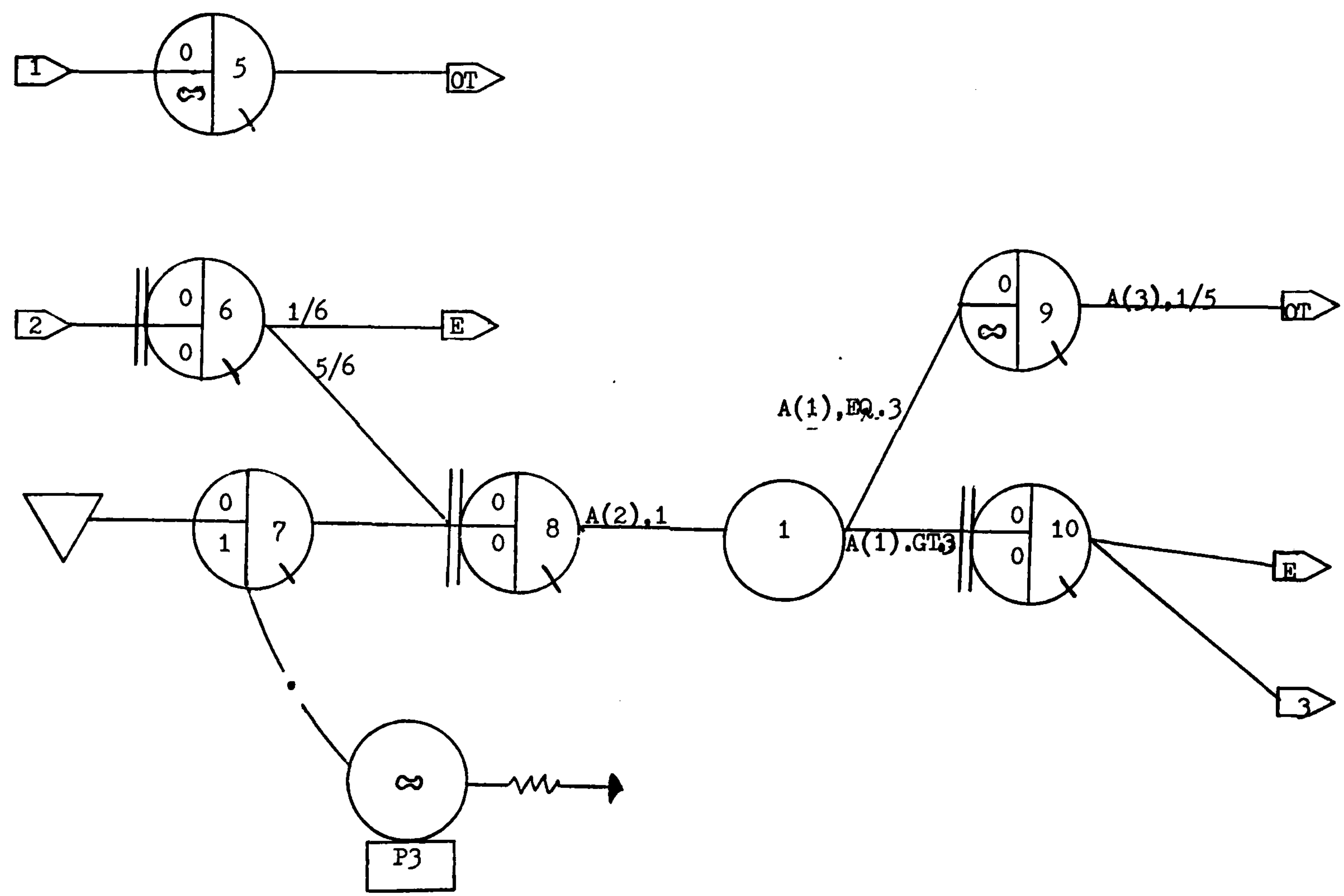
Circuit Switched Simulation Model

The circuit switched simulation model calculates delay time and throughput for the voice transmission of the Circuit switched voice/Packet switched data network shown in Figures 1 and 2. Presented in this appendix is the flowchart and the SLAM language simulation code for analyzing the circuit switched voice path. The key attributes are

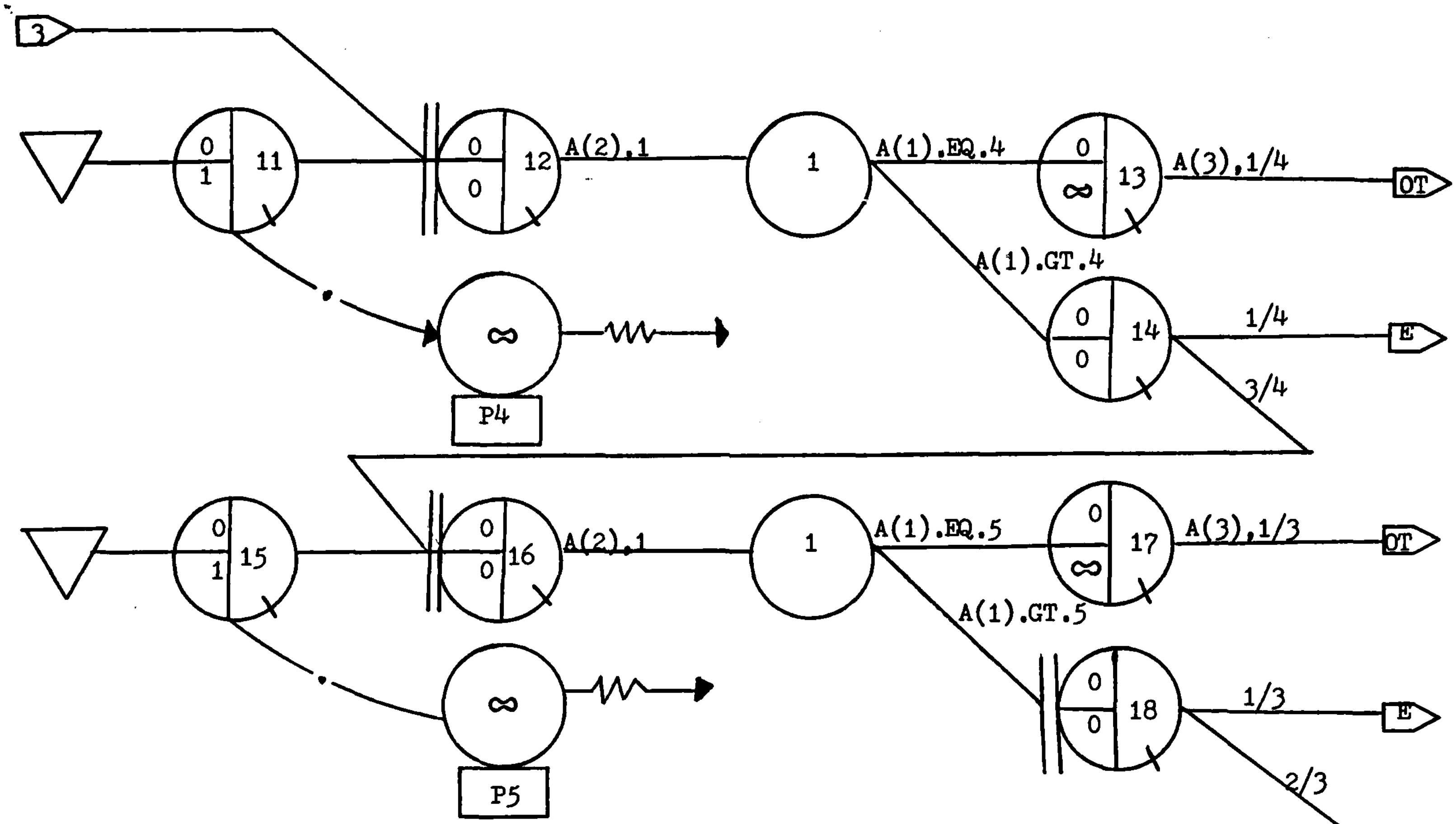
```
Attribute(1)= Destination Node
Attribute(2)= Service Rate for Reservation Signal
Attribute(3)= Service Rate for Message Transmission Signal
                                Transmission Request Signal
                                Channel Release Signal
Attribute(4)= Source Node
```



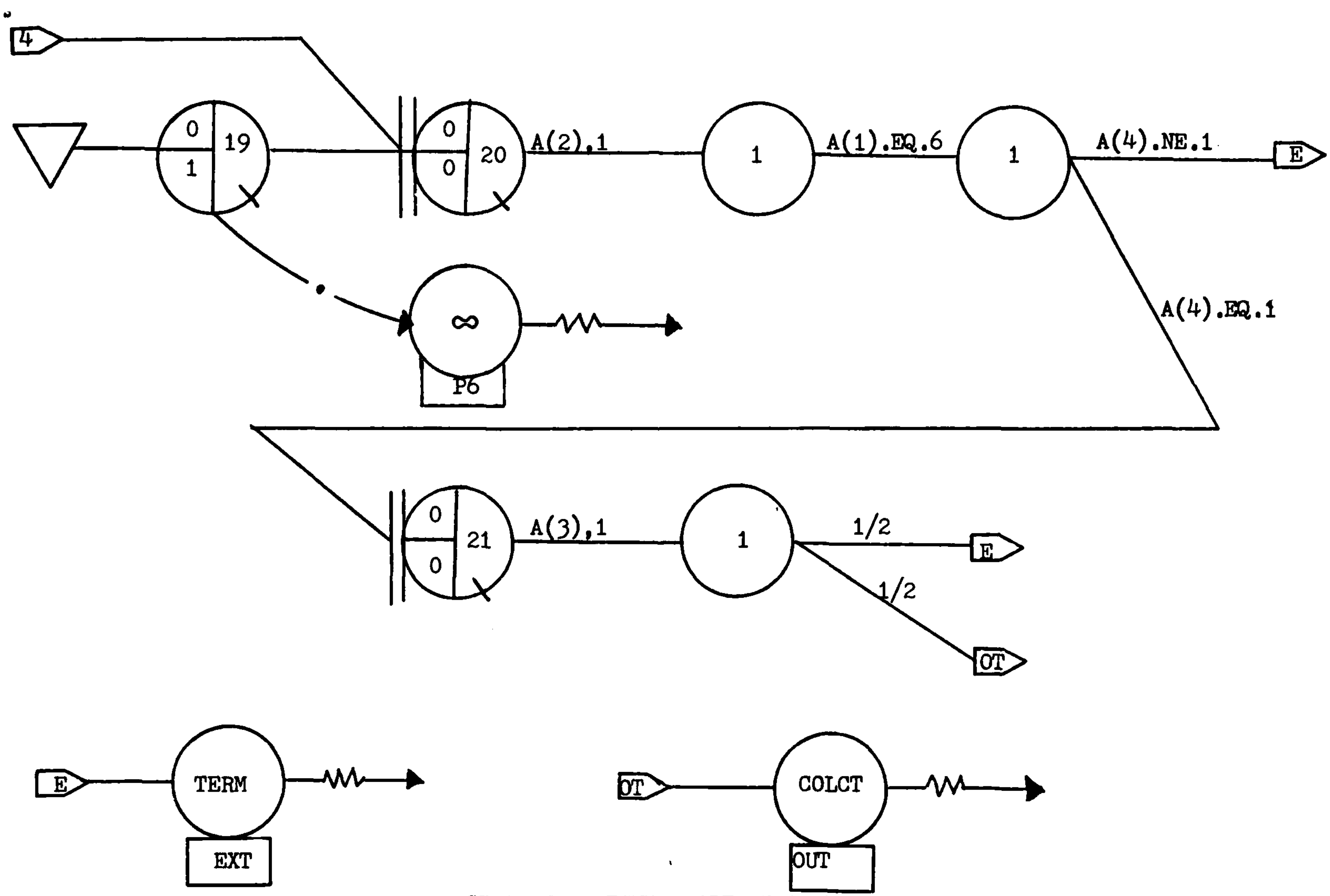
CIRCUIT SWITCHED MODEL FLOWCHART



CIRCUIT SWITCHED MODEL FLOWCHART



CIRCUIT SWITCHED MODEL FLOWCHART



CIRCUIT SWITCHED MODEL FLOWCHART

*PROGRAM FUNCTION: This simulation model represents a circuit switched network for the transmission of analog voice, by reserving the entire path prior to transmission of the message. The entire path is not released until the message is completed.

*ADJUSTABLE PARAMETERS: Arrival rates, source/destination nodes, queue lengths, block/balk specifications, traffic intensity and service rate.

```
GEN,SWALKER,NETIA,7/25/85;
LIMITS,21,5,300;
NETWORK;
        CREATE,EXPON(0.05),,5;       *Arrival rate of voice packets
        ASSIGN,ATRIB(1)=6;           *Destination Node
        ASSIGN,ATRIB(2)=.0023;       *Service rate of reservation signal
        ASSIGN,ATRIB(3)=.0383;       *Service rate of other signals
        ASSIGN,ATRIB(4)=1;           *Source node
        ACT,,,B1;                    *Node for simulation arrivals
        CREATE,EXPON(0.05),,5;
        ASSIGN,ATRIB(1)=6;
        ASSIGN,ATRIB(2)=.0023;
        ASSIGN,ATRIB(3)=.0383;
        ASSIGN,ATRIB(4)=2;
        ACT,,,B2;
        CREATE,EXPON(0.05),,5;
        ASSIGN,ATRIB(1)=6;
        ASSIGN,ATRIB(2)=.0023;
        ASSIGN,ATRIB(3)=.0383;
        ASSIGN,ATRIB(4)=3;
        ACT,,,B3;
        CREATE,EXPON(0.05),,5;
        ASSIGN,ATRIB(1)=6;
        ASSIGN,ATRIB(2)=.0023;
        ASSIGN,ATRIB(3)=.0383;
        ASSIGN,ATRIB(4)=4;
        ACT,,,B4;
        CREATE,EXPON(0.05),,5;
        ASSIGN,ATRIB(1)=6;
        ASSIGN,ATRIB(2)=.0023;
        ASSIGN,ATRIB(3)=.0383;
        ASSIGN,ATRIB(4)=5;
        ACT,,,B5;
        CREATE,EXPON(0.05),,5;
        ASSIGN,ATRIB(1)=6;
        ASSIGN,ATRIB(2)=.0023;
        ASSIGN,ATRIB(3)=.0383;
        ASSIGN,ATRIB(4)=6;
        ACT,,,B6;
B1      QUEUE(1),01,BALK(P1);
* Balking used to determine the number of rejected packets.
        ACT/1,,1,Q1;
Q1      QUEUE(2),0,0,BLOCK;
* Blocking used to prevent packets from entering the system until a
```

```
transmission is completed.
        ACT/2,ATRIB(2),1,Q2;
* Channel reservation signal serviced and sent to the next node.
B2      QUEUE(3),0,1,BALK(P2);
        ACT/3,,1,Q2;
Q2      QUEUE(4),0,0,BLOCK;
        ACT/4,ATRIB(2);
        GOON,1;
        ACT,,ATRIB(1).EQ.2,O2;
* If the destination is node 2 then take this route.
        ACT,,ATRIB(1).GT.2,C2;
* If the destination is not node 2 then take this route.
O2      QUEUE(5);
        ACT/5,ATRIB(3),1/6,OUT;
* If destination is reached the message is serviced and 1/6 of the
entities are sent to the COLCT node for statistics to be collected.
C2      QUEUE(6),0,0,BLOCK;
        ACT/6.,5/6,Q3;
*If destination is not reached the channel reservation signal is
serviced and 5/6 of entities are sent to the next node.
        ACT/6,,1/6,EXT;
B3      QUEUE(7),0,1,BALK(P3);
        ACT/7,,1,Q3;
Q3      QUEUE(8),0,0,BLOCK;
        ACT/8,ATRIB(2);
        GOON,1;
        ACT,,ATRIB(1).EQ.3,O3;
        ACT,,ATRIB(1).GT.3,C3;
O3      QUEUE(9);
        ACT/9ATRIB(3),1/5,OUT;
C3      QUEUE(10),0,0,BLOCK;
        ACT/10,,4/5,Q4;
        ACT/10,,1/5,EXT;
B4      QUEUE(11),0,1,BALK(P4);
        ACT/11,,1,Q4;
Q4      QUEUE(12),0,0,BLOCK;
        ACT/12.ATRIB(2);
        GOON,1;
        ACT.,ATRIB(1).EQ.4,O4;
        ACT,,ATRIB(1).GT.4,C4;
O4      QUEUE(13);
        ACT/13,ATRIB(3),1/4,OUT;
C4      QUEUE(14),0,0,BLOCK;
        ACT/14,,3/4,Q5;
        ACT/14,,1/4,EXT;
B5      QUEUE(15),0,1,BALK(P5);
        ACT/15,,1,Q5;
Q5      QUEUE(16),0,0,BLOCK;
        ACT/16,ATRIB(2);
        GOON,1;
        ACT,,ATRIB(1).EQ.5,O5;
        ACT,,ATRIB(1).GT.5,C5;
O5      QUEUE(17);
```

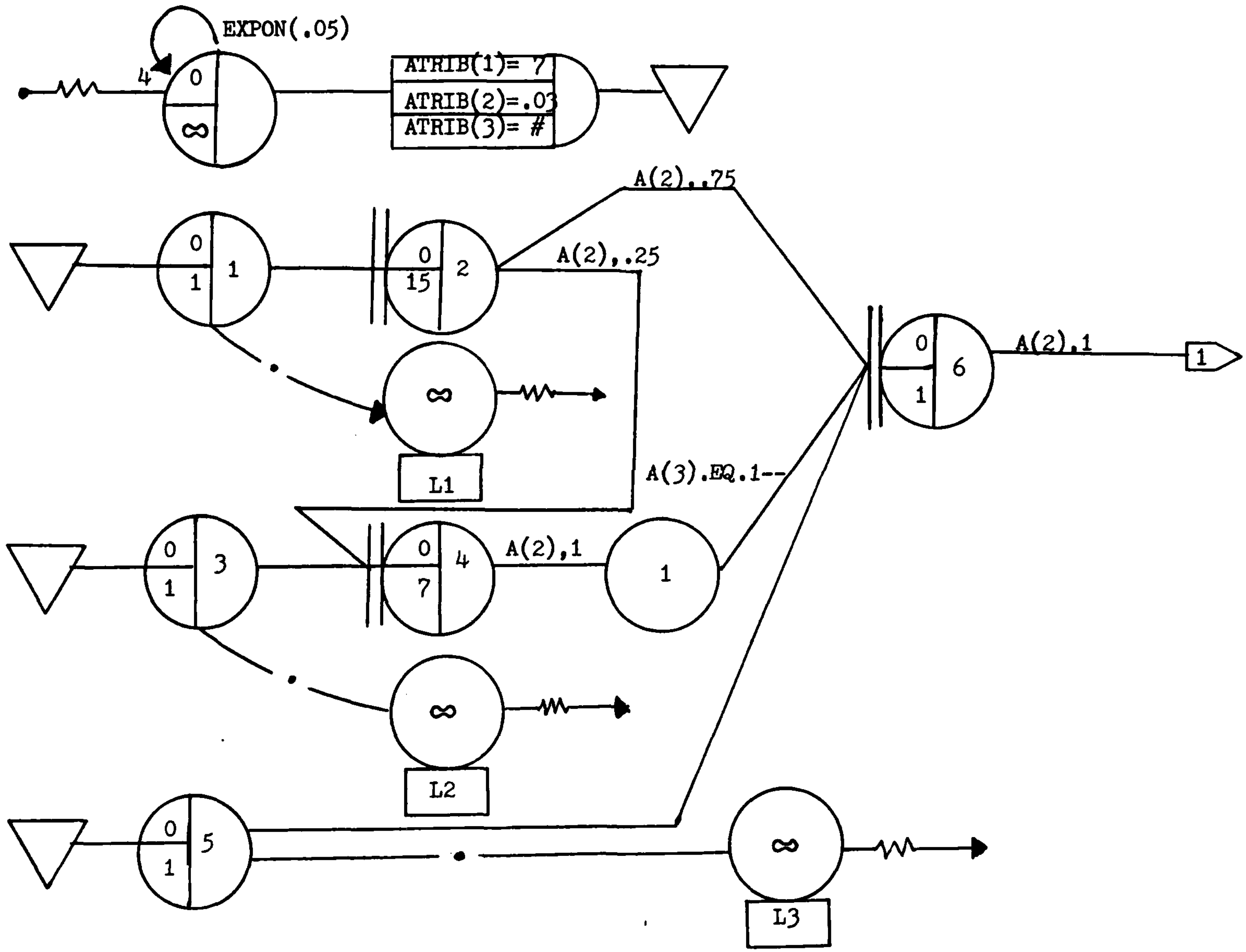
```
        ACT/17,ATRIB(3),1/3,OUT;
Q5      QUEUE(18),0,0,BLOCK;
        ACT/18,,2/3,Q6;
        ACT/18,,1/3,EXT;
B6      QUEUE(19),0,1,BALK(P6);
        ACT/19,,1,Q6;
Q6      QUEUE(20),0,0,BLOCK;
        ACT/20,ATRIB(2);
        GOON,1;
        ACT,,ATRIB(1).EQ.6,O6;
        ACT,,ATRIB(1).GT.6,EXT;
O6      GOON,1;
        ACT,,ATRIB(4).EQ.1,OT6;
* Only allows packets from Source node 1 to be processed. All others
are terminated.
        ACT,,ATRIB(4).NE.1,EXT;
OT6     QUEUE(21),0,0,BLOCK;
        ACT/21,ATRIB(3);
* Message transmission is serviced and channel opened for further
traffic.
        GOON,1;
        ACT,,1/2,OUT;
*Entities are sent to COLCT node for statistics. Only 1/2 of entities
are processed. The others are terminated.
        ACT,,1/2,EXT;
        TERM;
OUT     COLCT,INT(5),NODE TIME;
* This collect the statistics on delay time and throughput for
processed entities(packets).
        TERM;
P1      COLCT,BETWEEN,PLOSS1;
* This collects any packets that are rejected at the first node.
PLOSS1 denotes location where packet loss occurred.
        TERM;
P2      COLCT,BETWEEN,PLOSS2;
        TERM;
P3      COLCT,BETWEEN,PLOSS3;
        TERM;
P4      COLCT,BETWEEN,PLOSS4;
        TERM;
P5      COLCT,BETWEEN,PLOSS5;
        TERM;
P6      COLCT,BETWEEN,PLOSS6;
        TERM;
EXT     TERM;
* This is used to terminate excess entities.
        END;
INIT,0,100;
* Model run time of 100 time units.
FIN;
```

APPENDIX B

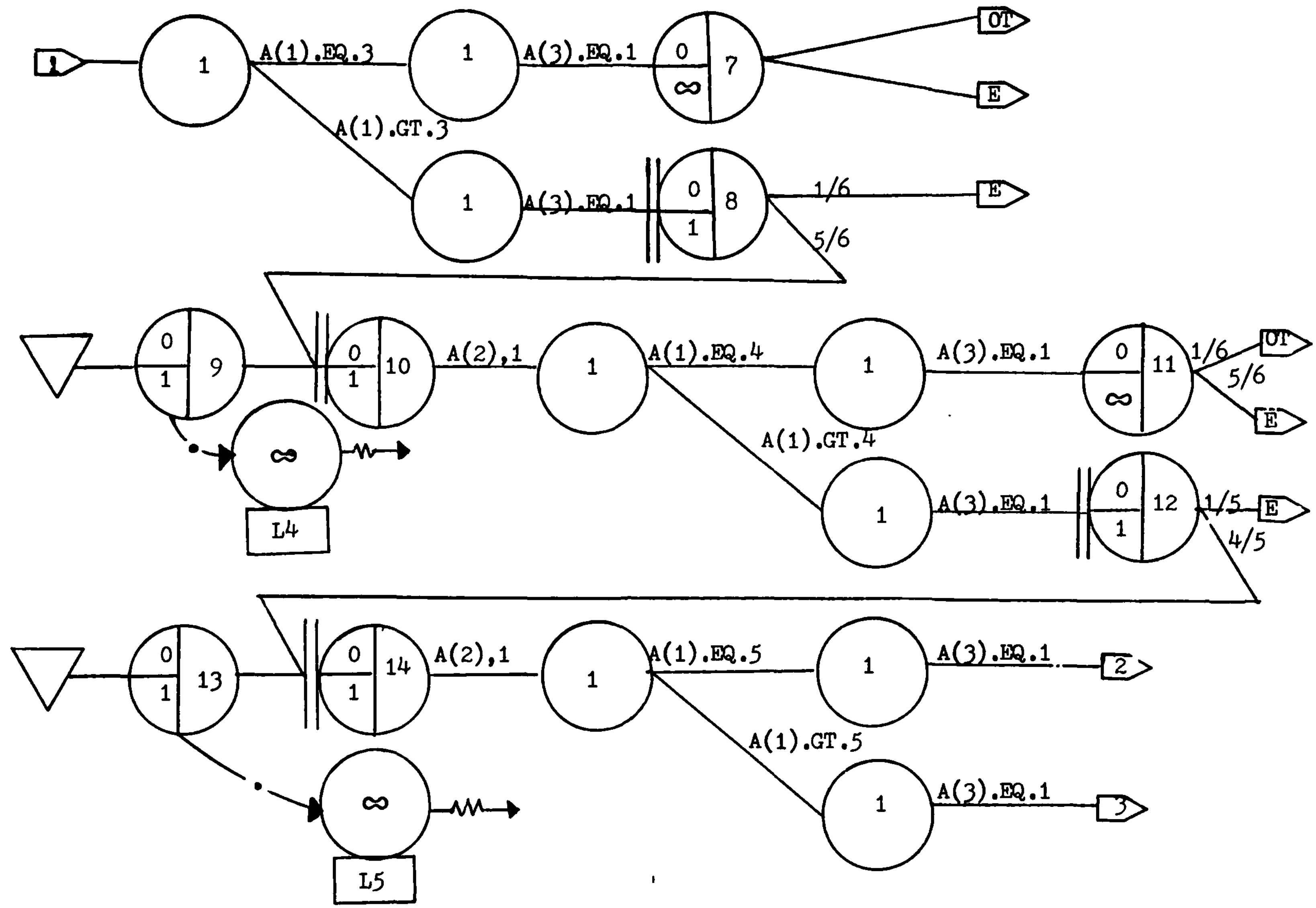
PACKET SWITCHED MODEL

Appendix B provides a flow chart and SLAM simulation code for analyzing a packet switched network with single arrivals per node. This is the simulation model used to calculate delay time and throughput for the data flow of the first network as shown in Figure and the bulk data of the third network shown in Figure 5. The key attributes are

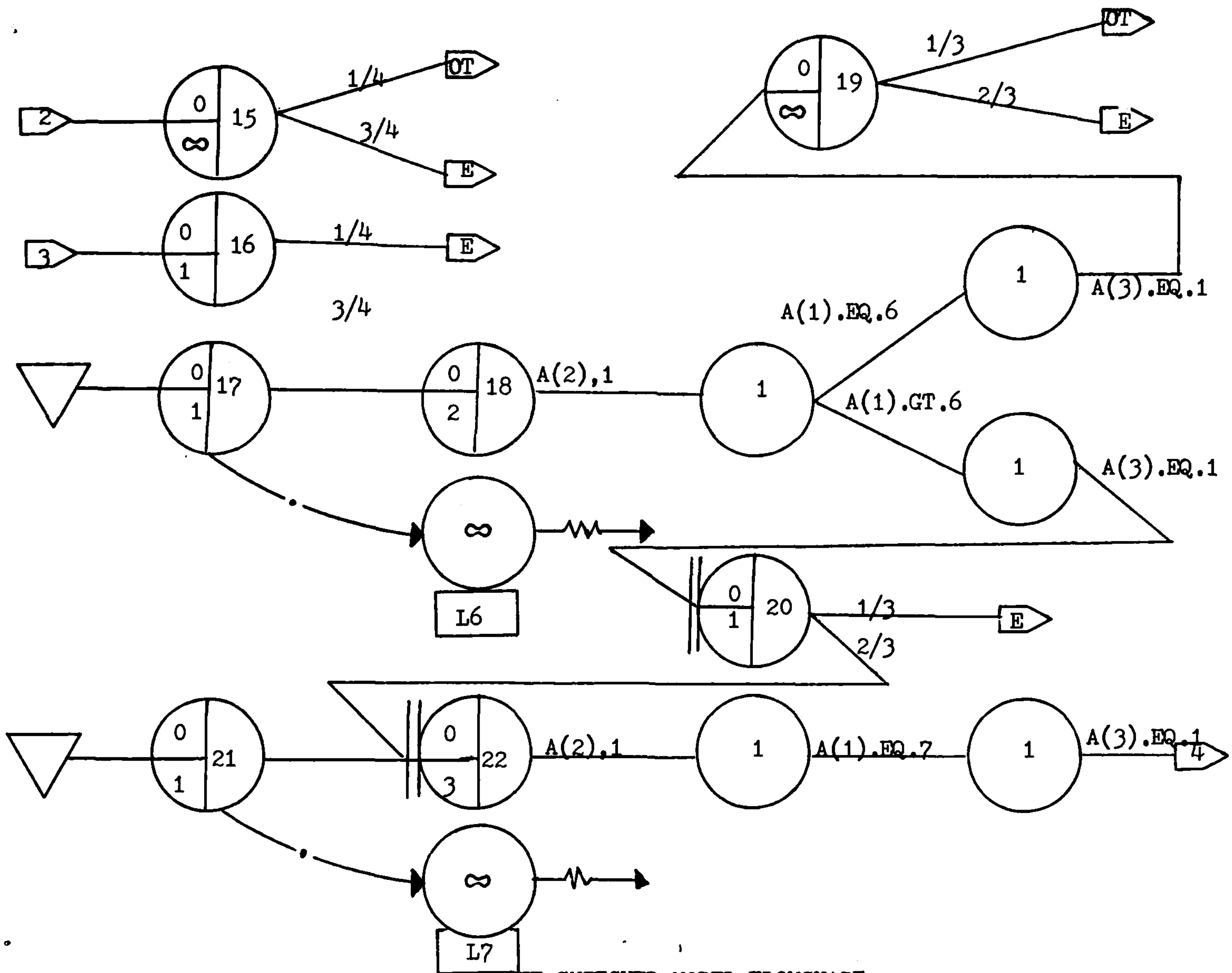
Attribute(1)= Source Designation Node
Attribute(2)= Service Rate
Attribute(3)= Destination Node



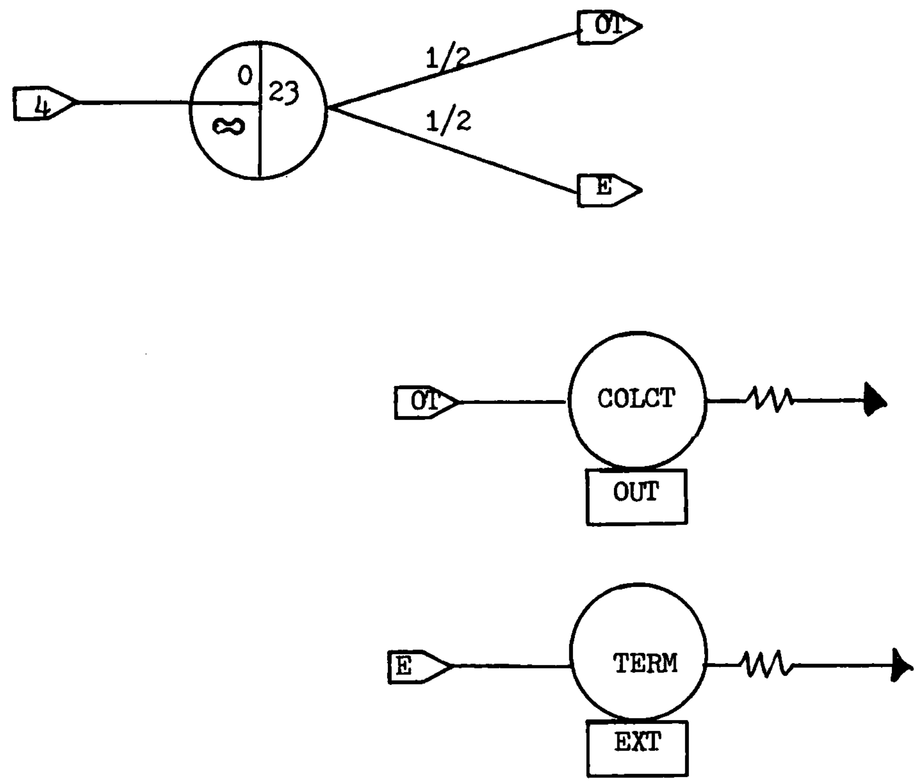
PACKET SWITCHED MODEL FLOWCHART



PACKET SWITCHED MODEL FLOWCHART



PACKET SWITCHED MODEL FLOWCHART



PACKET SWITCHED MODEL FLOWCHART

*PROGRAM FUNCTION: This simulation model represents the flow and outputs of a packet switched digital data network. Flow is based on source and destination node specification.

*ADJUSTABLE PARAMETERS: Arrival rates, source/destination nodes, queue lengths, block/balk specifications, path algorithm traffic intensity and service rates.

```
GEN,S.WALKER,NET13,7/19/85;
LIMITS,23,4,300;
NETWORK;
        CREATE,EXPON(.05),,4;     * Arrival rate and arrival code
        ASSIGN,ATRIB(1)=7;        * Destination node
        ASSIGN,ATRIB(2)=.003;     * Service rate
        ASSIGN,ATRIB(3)=1;        * Source node
        ACT,,,B1;                 * Node for simulation arrivals
        CREATE,EXPON(.05),,4;
        ASSIGN,ATRIB(1)=7;
        ASSIGN,ATRIB(2)=.003;
        ASSIGN,ATRIB(3)=2;
        ACT,,,B2;
        CREATE,EXPON(.05),,4;
        ASSIGN,ATRIB(1)=7;
        ASSIGN,ATRIB(2)=.003;
        ASSIGN,ATRIB(3)=3;
        ACT,,,B3;
        CREATE,EXPON(.05),,4;
        ASSIGN,ATRIB(1)=7;
        ASSIGN,ATRIB(2)=.003;
        ASSIGN,ATRIB(3)=4;
        ACT,,,B4;
        CREATE,EXPON(.05),,4;
        ASSIGN,ATRIB(1)=7;
        ASSIGN,ATRIB(2)=.003;
        ASSIGN,ATRIB(3)=5;
        ACT,,,B5;
        CREATE,EXPON(.05),,4;
        ASSIGN,ATRIB(1)=7;
        ASSIGN,ATRIB(2)=.003;
        ASSIGN,ATRIB(3)=6;
        ACT,,,B6;
B1      QUEUE(1),0,1,BALK;
* Balking used to determine the number of rejected.
        ACT/1,,,Q1;
Q1      QUEUE(2),0,15,BLOCK;
* Blocking used to prevent packets from entering the system until
buffer capacity is available. If buffer space is not immediately
available the packet will balk(reject). 15 is the maximum number
allowed in the queue at any one time, see Table 3.
        ACT/2,ATRIB(2),0.75,Q3;
* Path algorithm percentage used to 75 % of packets over shortest
path to destination and 25% of packets over a path with one
additional node in the path.
```

```
        ACT/2,ATRIB(2),0.25,Q2;
B2      QUEUE(3),0,1,BALK(L2);
        ACT/3,,,Q2;
Q2      QUEUE(4),0,7,BLOCK;
        ACT/4,ATRIB(2);
        GOON,1;
        ACT,,ATRIB(3).EQ.1,B3;
B3      QUEUE(5),0,1,BALK(L3);
        ACT/5,,,Q3;
Q3      QUEUE(6),0,1,BLOCK;
* This is the first node for possible output.
        ACT/6,ATRIB(2);
        GOON,1;
        ACT,,ATRIB(1).EQ.3,O3;
* If this is the destination then entities are sent to O3.
        ACT,,ATRIB(1).GT.3,C3;
* If this is not destination then entities continued on through the
network.
O3      GOON,1;
        ACT,,ATRIB(3).EQ.1,OT3;
* Only entities coming from node 1 are allowed to process.
OT3     QUEUE(7);
        ACT/7,,1/6,OUT;
        ACT/7,,5/6,EXT;
* 1/6 of entities are processed if this is the destination node and
5/6 of entities are terminated.
C3      GOON,1;
        ACT,,ATRIB(3).EQ.1,CT3;
* If destination is not reached entities continue on to next queue.
CT3     QUEUE(8),0,1,BLOCK;
        ACT/8,,5/6,Q4;
        ACT/8,,1/6,EXT;
* Only 5/6 of entities are allowed to continue to next queue and 1/6
of entities are terminated.
B4      QUEUE(9),0,1,BALK(L4);
        ACT/9,,,Q4;
Q4      QUEUE(10),0,1,BLOCK;
        ACT/10,ATRIB(2);
        GOON,1;
        ACT,,ATRIB(1).EQ.4,O4;
        ACT,,ATRIB(1).GT.4,C4;
O4      GOON,1;
        ACT,,ATRIB(3).EQ.1,OT4;
OT4     QUEUE(11);
        ACT/11,,1/5,OUT;
        ACT/11,,4/5,EXT;
C4      GOON,1;
        ACT,,ATRIB(3).EQ.1,CT4;
CT4     QUEUE(12),0,1,BLOCK;
        ACT/12,,4/5,Q5;
        ACT/12,,1/5,EXT;
B5      QUEUE(13),0,1,BALK(L5);
        ACT/13,,,Q5;
```

```
Q5      QUEUE(14),0,1,BLOCK;
        ACT/14,ATRIB(2);
        GOON,1;
        ACT,,ATRIB(1).EQ.5,O5;
        ACT,,ATRIB(1).GT.5,C5;
O5      GOON,1;
        ACT,,ATRIB(3).EQ.1,OT5;
OT5     QUEUE(15);
        ACT/15,,1/4,OUT;
        ACT/15,,3/4,EXT;
C5      GOON,1;
        ACT,,ATRIB(3).EQ.1,CT5;
CT5     QUEUE(16),0,1,BLOCK;
        ACT/16,,3/4,Q6;
        ACT/16,,1/4,EXT;
B6      QUEUE(17),0,1,BALK(L6);
        ACT/17,,,Q6;
Q6      QUEUE(18),0,2,BLOCK;
        ACT/18,ATRIB(2);
        GOON,1;
        ACT,,ATRIB(1).EQ.6,O6;
        ACT,,ATRIB(1).GT.6,C6;
O6      GOON,1;
        ACT,,ATRIB(3).EQ.1,OT6;
OT6     QUEUE(19);
        ACT/19,,1/3,OUT;
        ACT/19,,2/3.EXT;
C6      GOON,1;
        ACT,,ATRIB(3).EQ.1,CT6;
CT6     QUEUE(20),0,1,BLOCK;
        ACT/20,,2/3,Q7;
        ACT/20,,1/3,EXT;
B7      QUEUE(21),0,1,BALK(L7);
        ACT/21,,,Q7;
Q7      QUEUE(22),0,3,BLOCK;
        ACT/22.ATRIB(2);
        GOON,1;
        ACT,,ATRIB(1).EQ.7,O7;
O7      GOON,1;
        ACT,,ATRIB(3).EQ.1,OT7;
* Only entities sent from node are processed.
OT7     QUEUE(23);
        ACT/23,,1/2,OUT;
        ACT/23,,1/2,EXT;
* 1/2 of entities allowed processing are sent to the collect node
and 1/2 are terminated.
        TERM;
OUT     COLCT,INT(4),NODE TIME;
* This node collects statistics on delay time and throughput for
processed entities which have an arrival code of 4.
        TERM;
L1      COLCT,BETWEEN,PKT LOSS;
* Rejected packets are counted here for each node.
```

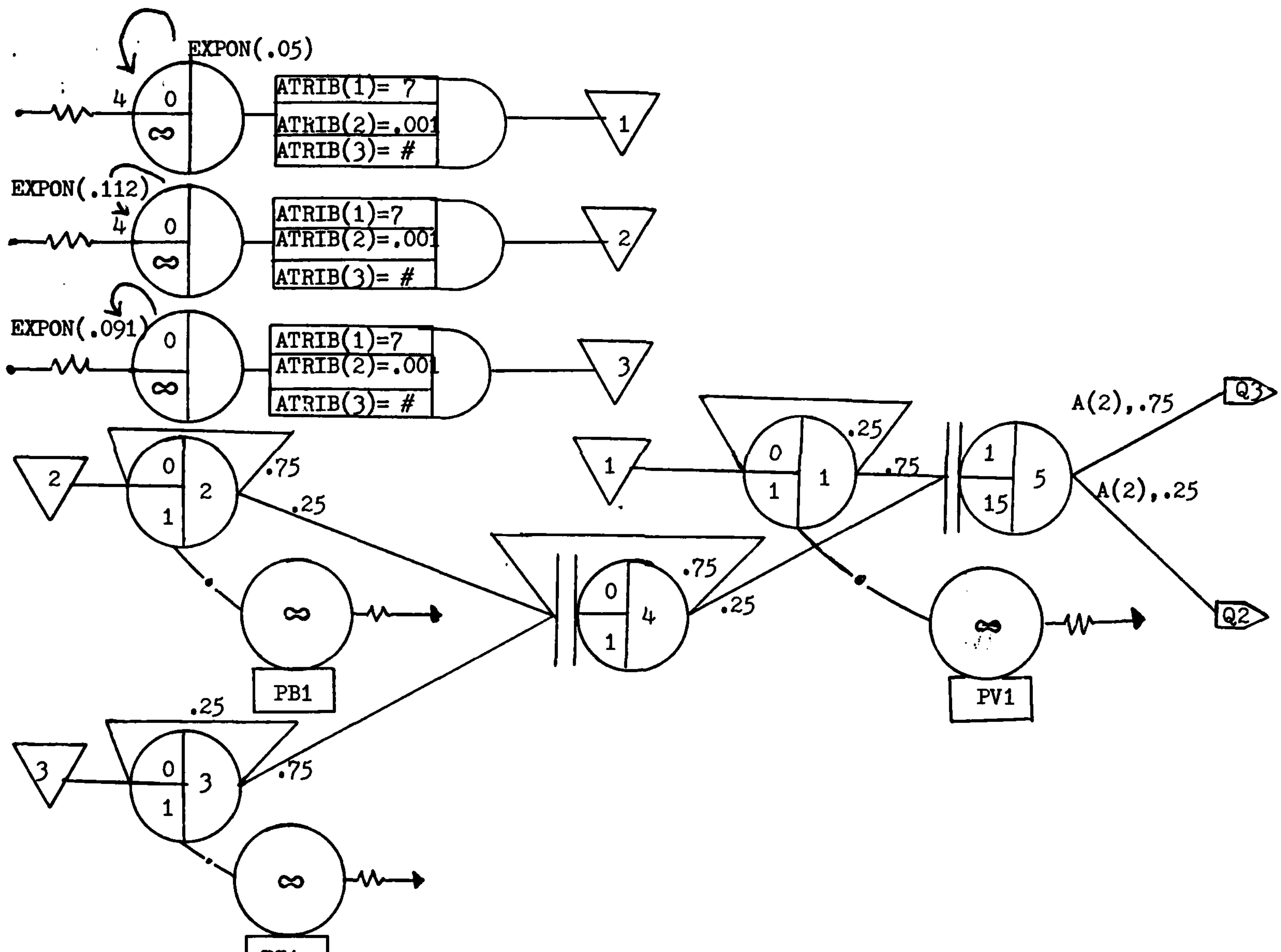
```
        TERM;
L2      COLCT,BETWEEN,PKT LOSS;
        TERM;
L3      COLCT,BETWEEN,PKT LOSS;
        TERM;
L4      COLCT,BETWEEN,PKT LOSS;
        TERM;
L5      COLCT,BETWEEN,PKT LOSS;
        TERM;
L6      COLCT,BETWEEN,PKT LOSS;
        TERM;
L7      COLCT,BETWEEN,PKT LOSS;
        TERM;
EXT     TERM;
* Terminate node for excess entities.
        TERM;
        END;
INIT,0,100;
* System run time for simulation model.
FIN;
```

APPENDIX C

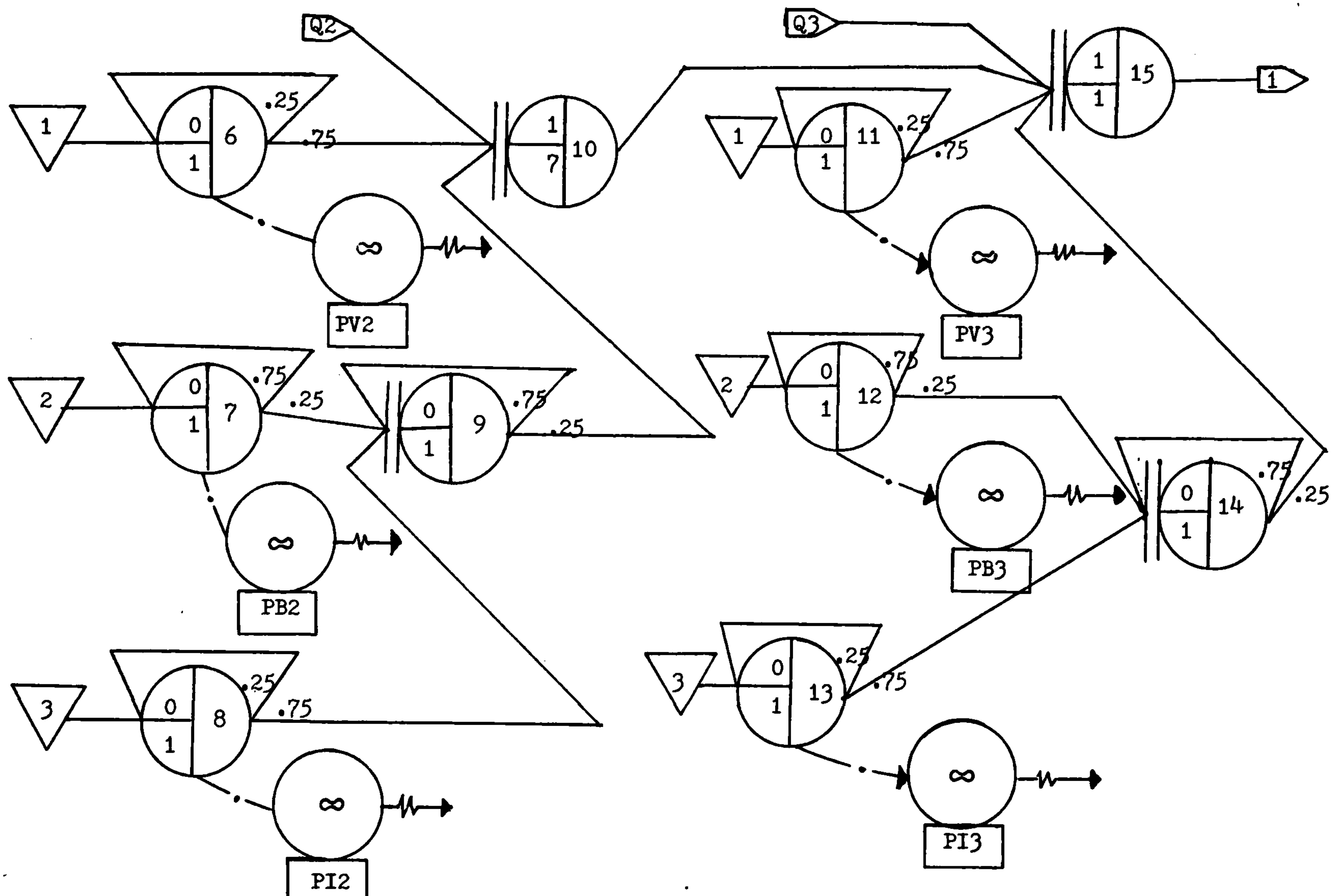
MULTIPLE ARRIVAL PACKET SWITCHED MODEL

This appendix provides the flowchart and SLAM simulation code for analysis of multiple arrival packet switched networks. This particular code was used to analyze the digitized voice, interactive data and bulk data network. A modification of this simulation code, eliminating the bulk data arrivals, allows use of this simulation code for analysis of the digitized voice and interactive portion of the third network. The attributes are

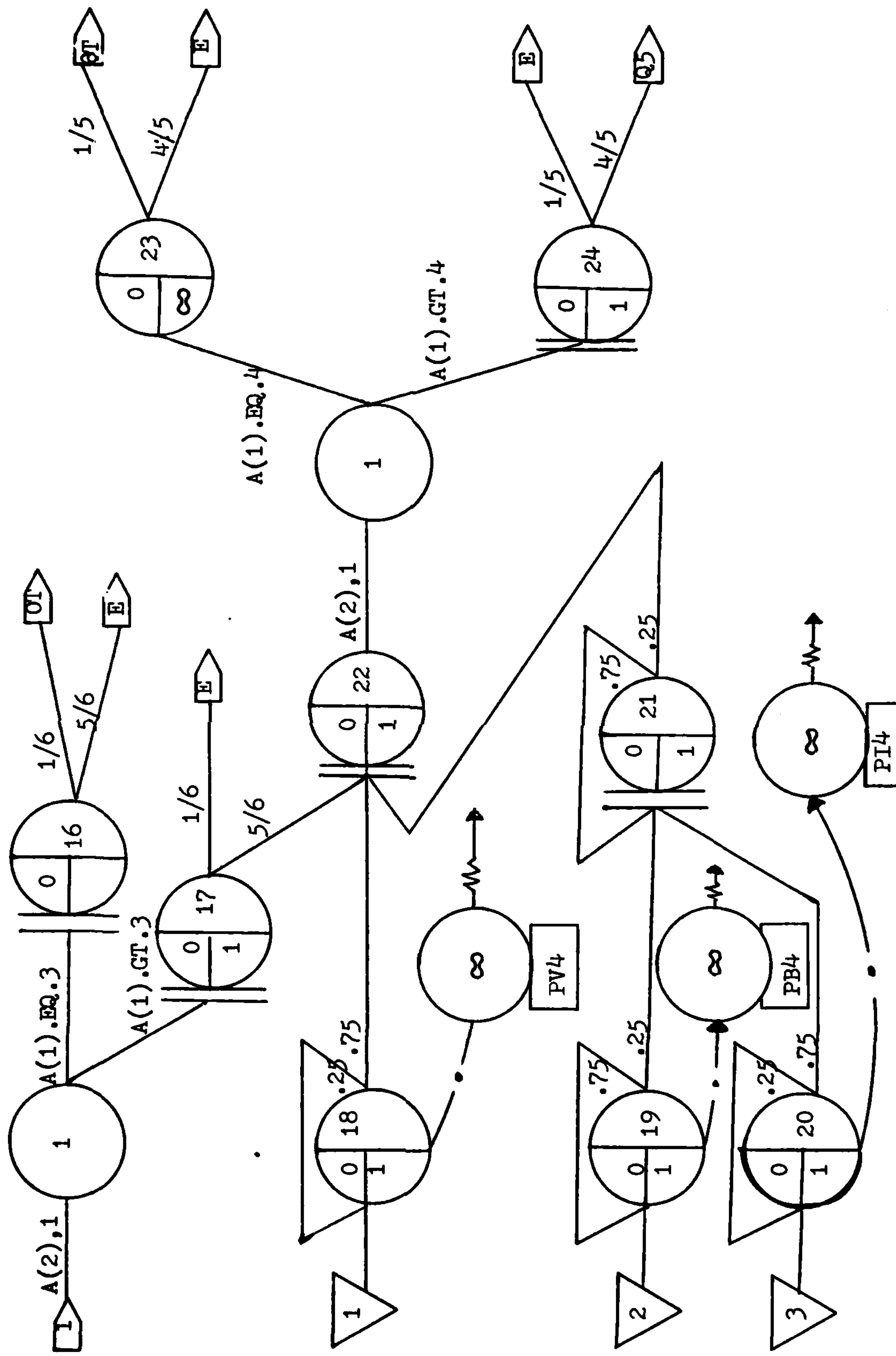
Attribute(1)= Source Designation Node
Attribute(2)= Service Rate
Attribute(3)= Destination Node



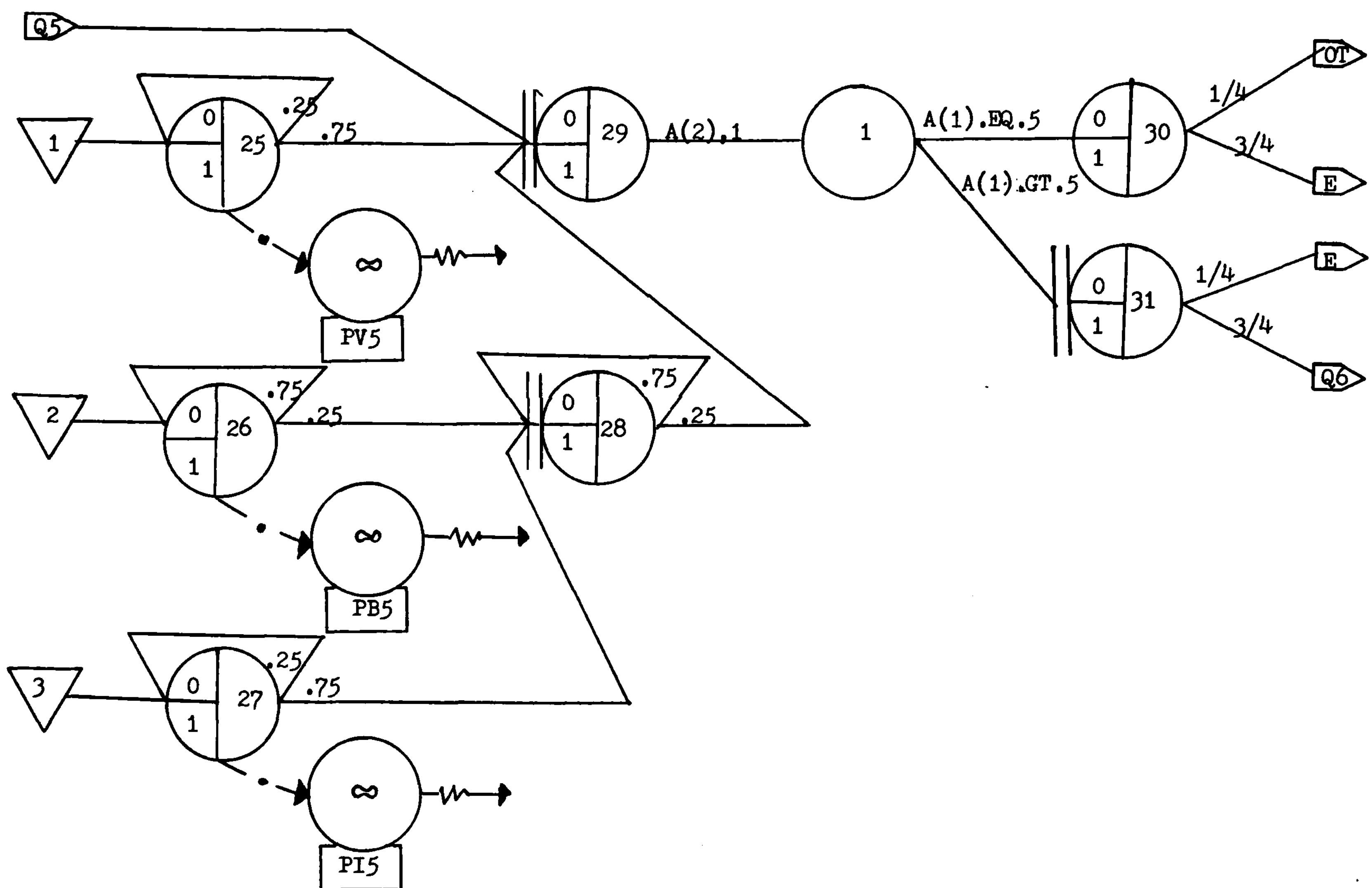
MULTIPLE ARRIVAL PACKET SWITCHED MODEL FLOWCHART



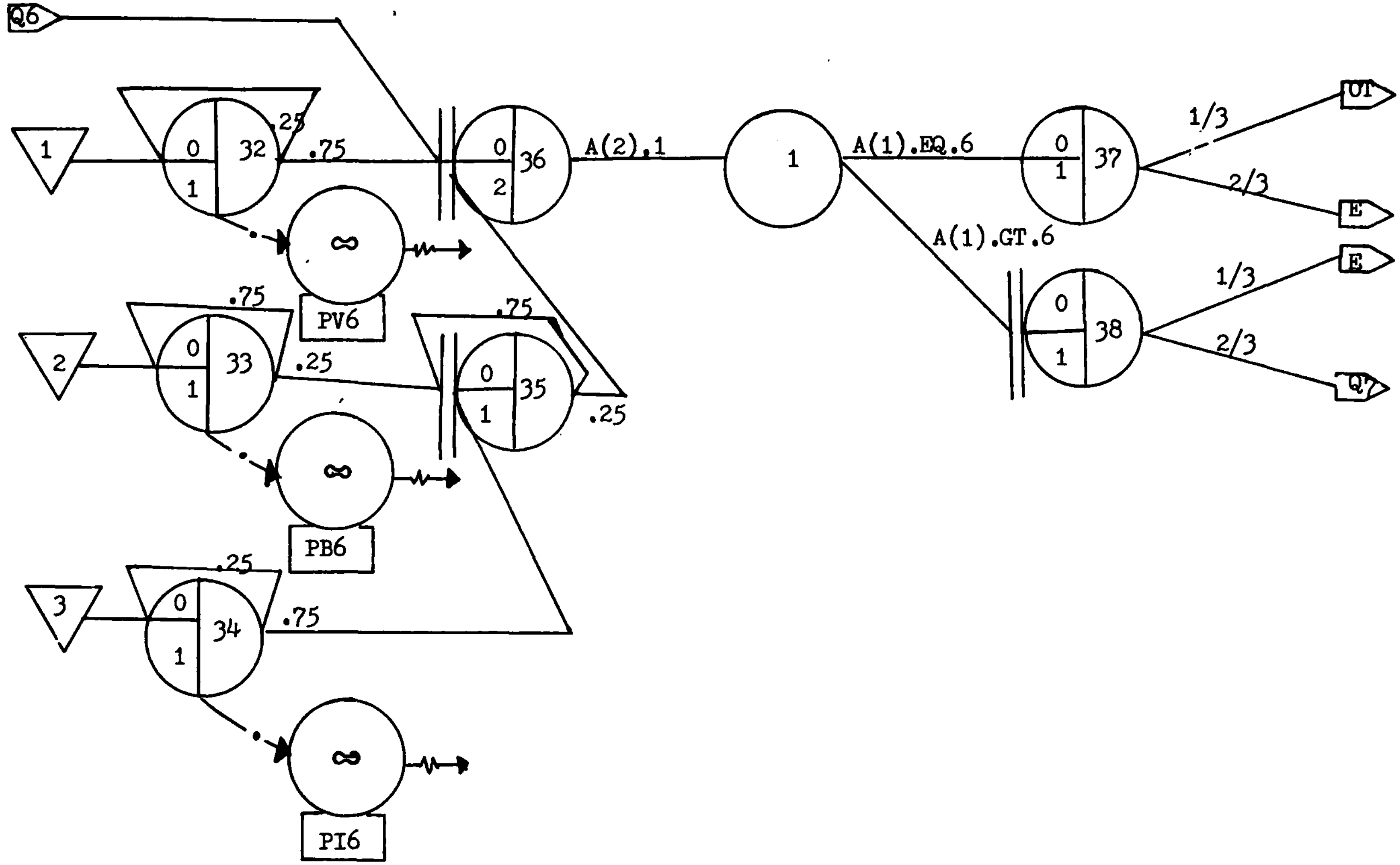
MULTIPLE ARRIVAL PACKET SWITCHED MODEL FLOWCHART



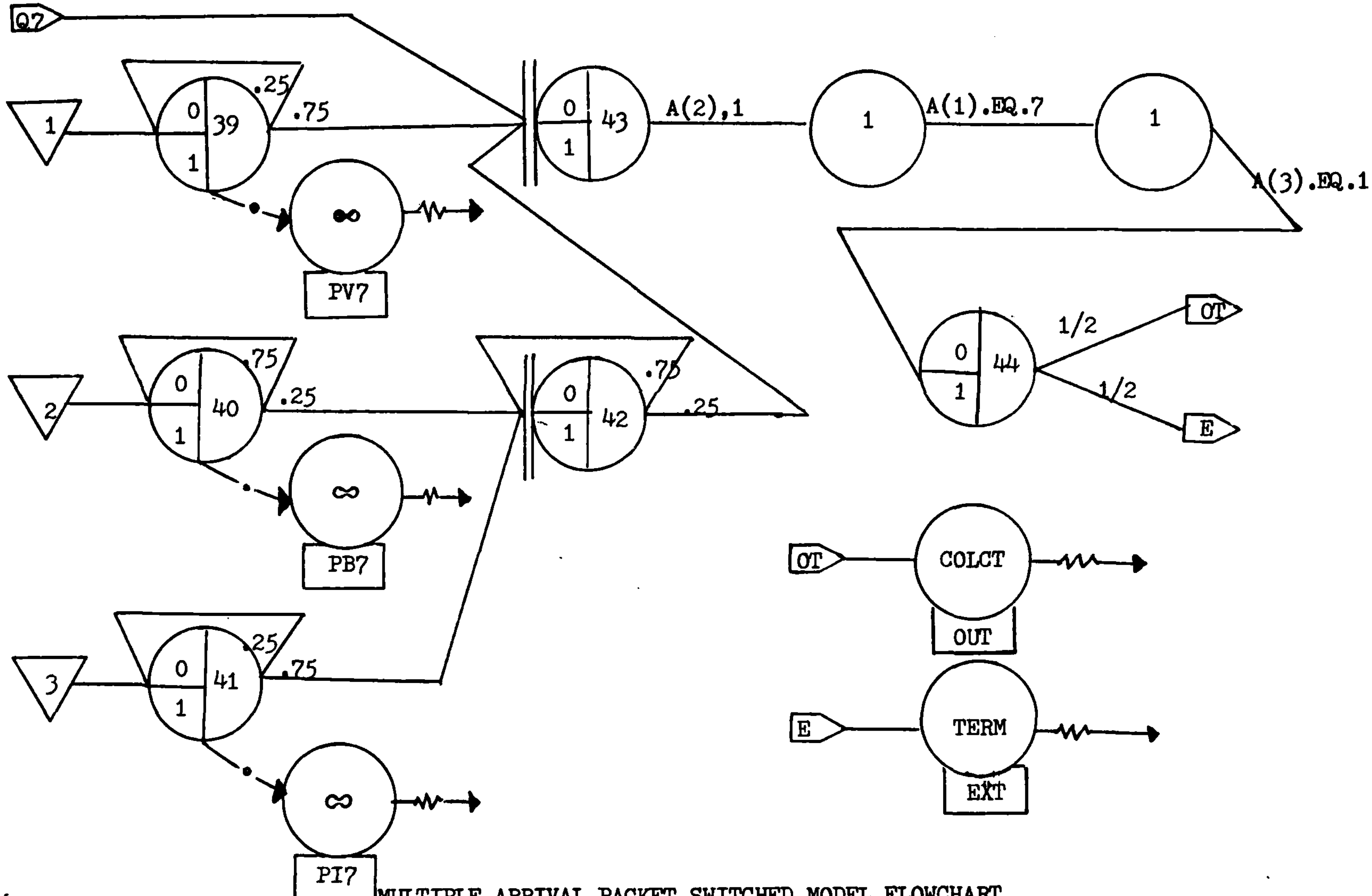
MULTIPLE ARRIVAL PACKET SWITCHED MODEL FLOWCHART



MULTIPLE ARRIVAL PACKET SWITCHED MODEL FLOWCHART



MULTIPLE ARRIVAL PACKET SWITCHED MODEL FLOWCHART



MULTIPLE ARRIVAL PACKET, SWITCHED MODEL FLOWCHART

*PROGRAM FUNCTION: This simulation model represents a packet switched network which transmits integrated digital voice and data. This is an example of a multiple arrival packet switched network.

*ADJUSTABLE PARAMETERS: Arrival rates, source/destination nodes, queue length, block/balk specifications, path algorithm, traffic intensity, percentages of voice and data, percentages of interactive and bulk data and service rates.

```
GEN,SWALKER,NET2I,5/30/85;
LIMITS,44,4,300;
NETWORK;
          CREATE,EXPON(.05),,4;    * Arrival rate for voice and arrival
                                     code
          ASSIGN,ATRIB(1)=7;       * Destination node
          ASSIGN,ATRIB(2)=.001;    * Service rate
          ASSIGN,ATRIB(3)=1;       * Source node
          ACT,,,V1;                * Node for voice simulation arrivals
          CREATE,EXPON(.112),,4;   * Arrival rate for bulk data and
                                     arrival code
          ASSIGN,ATRIB(1)=7;
          ASSIGN,ATRIB(2)=.001;
          ASSIGN,ATRIB(3)=1;
          ACT,,,B1;                * Node for bulk simulation arrivals
          CREATE,EXPON(.091),,4;   * Arrival rate for interactive data
                                     and arrival code
          ASSIGN,ATRIB(1)=7;
          ASSIGN,ATRIB(2)=.001;
          ASSIGN,ATRIB(3)=1;
          ACT,,,I1;                * Node for interactive simulation
                                     arrivals
          CREATE,EXPON(.05),,4;
          ASSIGN,ATRIB(1)=7;
          ASSIGN,ATRIB(2)=.001;
          ASSIGN,ATRIB(3)=2;
          ACT,,,V2;
          CREATE,EXPON(.112),,4;
          ASSIGN,ATRIB(1)=7;
          ASSIGN,ATRIB(2)=.001;
          ASSIGN,ATRIB(3)=2;
          ACT,,,B2;
          CREATE,EXPON(.091),,4;
          ASSIGN,ATRIB(1)=7;
          ASSIGN,ATRIB(2)=.001;
          ASSIGN,ATRIB(3)=2;
          ACT,,,I2;
          CREATE,EXPON(.05),,4;
          ASSIGN,ATRIB(1)=7;
          ASSIGN,ATRIB(2)=.001;
          ASSIGN,ATRIB(3)=3;
          ACT,,,V3;
          CREATE,EXPON(.112),,4;
          ASSIGN,ATRIB(1)=7;
```

```
ASSIGN,ATRIB(2)=.001;
ASSIGN,ATRIB(3)=3;
ACT,,,B3;
CREATE,EXPON(.091),,4;
ASSIGN,ATRIB(1)=7;
ASSIGN,ATRIB(2)=.001;
ASSIGN,ATRIB(3)=3;
ACT,,,I3;
CREATE,EXPON(.05),,4;
ASSIGN,ATRIB(1)=7;
ASSIGN,ATRIB(2)=.001;
ASSIGN,ATRIB(3)=4;
ACT,V4;
CREATE,EXPON(112),,4;
ASSIGN,ATRIB(1)=7;
ASSIGN,ATRIB(2)=.001;
ASSIGN,ATRIB(3)=4;
ACT,,,B4;
CREATE,EXPON(.091),,4;
ASSIGN,ATRIB(1)=7;
ASSIGN,ATRIB(2)=.001;
ASSIGN,ATRIB(3)=4;
ACT,,,I4;
CREATE,EXPON(.05),,4;
ASSIGN,ATRIB(1)=7;
ASSIGN,ATRIB(2)=.001;
ASSIGN,ATRIB(3)=5;
ACT,,,V5;
CREATE,EXPON(.112),,4;
ASSIGN,ATRIB(1)=7;
ASSIGN,ATRIB(2)=.001;
ASSIGN,ATRIB(3)=5;
ACT,,,B5;
CREATE,EXPON(.091),,4;
ASSIGN,ATRIB(1)=7;
ASSIGN,ATRIB(2)=.001;
ASSIGN,ATRIB(3)=5;
ACT,,,I5;
CREATE,EXPON(.05),,4;
ASSIGN,ATRIB(1)=7;
ASSIGN,ATRIB(2)=.001;
ASSIGN,ATRIB(3)=6;
ACT,,,V6;
CREATE,EXPON(.112),,4;
ASSIGN,ATRIB(1)=7;
ASSIGN,ATRIB(2)=.001;
ASSIGN,ATRIB(3)=6;
ACT,,,B6;
CREATE,EXPON(.091),,4;
ASSIGN,ATRIB(1)=7;
ASSIGN,ATRIB(2)=.001;
ASSIGN,ATRIB(3)=6;
ACT,,,I6;
```

```
        CREATE,EXPON(.05),,4;
        ASSIGN,ATRIB(1)=7;
        ASSIGN,ATRIB(2)=.001;
        ASSIGN,ATRIB(3)=7;
        ACT,,,V7;
        CREATE,EXPON(.112),,4;
        ASSIGN,ATRIB(1)=7;
        ASSIGN,ATRIB(2)=.001;
        ASSIGN,ATRIB(3)=7;
        ACT,,,B7;
        CREATE,EXPON(.091),,4;
        ASSIGN,ATRIB(1)=7;
        ASSIGN,ATRIB(2)=.001;
        ASSIGN,ATRIB(3)=7;
        ACT,,,I7;
V1      QUEUE(1),0,1,BALK(PV1);
* First voice arrival node.  Balking used to collect the number of
rejected voice packets.
        ACT/1,,.75,Q1;
        ACT/1,,.25,V1;
* Allocates the percentage of voice allowed into the system.  Those
entities not allowed into the system the system are rerouted back
through the voice node.
B1      QUEUE(2),0,1,BALK(PB1);
* First bulk data arrival node.
        ACT/2,,.25,D1;
        ACT/2,,.75,B1;
* Allocates the percentage of bulk data allowed into the system.
I1      QUEUE(3),0,1,BALK(PI1);
* First Interactive data arrival node.
        ACT/3,,.75,D1;
        ACT/3,,.25,I1;
* Allocates the percentage of interactive data allowed into the
system.
D1      QUEUE(4),0,1,BLOCK;
* This provides the blocking for bulk and interactive data arrivals.
If buffer capacity is not available a packet is rejected.
        ACT/4,,.25,Q1;
        ACT/4,,.75,D1;
* This allocates the percentage of total data into the system.
Q1      QUEUE(5),1,15,BLOCK;
* This node blocks voice and total data packets is the buffer
capacity is full.  The maximum queue length is 15 per Table 3.
        ACT/5,ATRIB(2),0.75,Q3;
        ACT/5,ATRIB(2),0.25,Q2;
* Path algorithm percentages.  75% take the shortest path and 25%
take the route with one additional node in it.
V2      QUEUE(6),0,1,BALK(PV2);
        ACT/6,,.75,Q2;
        ACT/6,,.25,V2;
B2      QUEUE(7),0,1,BALK(PB2);
        ACT/7,,.25,D2;
        ACT/7,,.75,B2;
```

```
I2       QUEUE(8),0,1,BALK(PI2);
         ACT/8,,.75,D2;
         ACT/8,,.25,I2;
D2       QUEUE(9),0,01,BLOCK;
         ACT/9,,.25,Q2;
         ACT/9,,.75,D2;
Q2       QUEUE(10),1,7,BLOCK;
         ACT/10,ATRIB(2),1,V3;
V3       QUEUE(11),0,1,BALK(PV3);
         ACT/11,,.75,B3;
         ACT/11,,.25,V3;
B3       QUEUE(12),0,1,BALK(PB3);
         ACT/12,,.25,D3;
         ACT/12,,.75,B3;
I3       QUEUE(13),0,1,BALK(PI3);
         ACT/13,,.75,D3;
         ACT/13,,.25,I3;
D3       QUEUE(14),0,1,BLOCK;
         ACT/14,,.25,Q3;
         ACT/14,,.75,D3;
Q3       QUEUE(15),1,1,BLOCK;
         ACT/15,ATRIB(2);
         GOON,1;
         ACT,,ATRIB(1).EQ.3,O3;
         ACT,,ATRIB(1).GT.3,C3;
* Packet is serviced and sent to the appropriate location.  If
destination is this node then O3 else the packet continues through
the system.
O3       QUEUE(16);
         ACT/16,,1/6,OUT;
         ACT/16,,5/6,EXT;
* If packet has reached destination then 1/6 of entities are output
to the collect.  5/6 of entities are terminated.
C3       QUEUE(17),0,1,BLOCK;
         ACT/17,,5/6,Q4;
         ACT/17,,1/6,EXT;
* If packet has not reached destination then 5/6 of entities continue
and 1/6 of entities are terminated.
V4       QUEUE(18),0,1,BALK(PV4);
         ACT/18,,.75,Q4;
         ACT/18,,.25,V4;
B4       QUEUE(19),0,1,BALK(PB4);
         ACT/19,,.25,D4;
         ACT/19,,.75,B4;
I4       QUEUE(20),0,1,BALK(PI4);
         ACT/20,,75,D4;
         ACT/20,,.25,I4;
D4       QUEUE(21),0,1,BLOCK;
         ACT/21,,.25,Q4;
         ACT/21,,.75,D4;
Q4       QUEUE(22),0,1,BLOCK;
         ACT/22,ATRIB(2);
         GOON,1;
```

```
        ACT,,ATRIB(1).EQ.4,O4;
        ACT,,ATRIB(1).GT.4,C4;
O4      QUEUE(23);
        ACT/23,,1/5,OUT;
        ACT/23,,4/5,EXT;
C4      QUEUE(24),0,1,BLOCK;
        ACT/24,,4/5,Q5;
        ACT/24,,1/5,EXT;
V5      QUEUE(25),0,1,BALK(PV5);
        ACT/25,,.75,Q5;
        ACT/25,,.25,V5;
B5      QUEUE(26),0,1,BALK(PB5);
        ACT/26,,.25,D5;
        ACT/26,,.75,B5;
I5      QUEUE(27),0,1,BALK(PK5);
        ACT/27,,.75,D5;
        ACT/27,,.25,I5;
D5      QUEUE(28),0,1,BLOCK;
        ACT/28,,.25,Q5;
        ACT/28,,.75,D5;
Q5      QUEUE(29),0,1,BLOCK;
        ACT/29,ATRIB(2);
        GOON,1;
        ACT,,ATRIB(1),EQ.5,O5;
        ACT,,ATRIB(1).GT.5,C5;
O5      QUEUE(30);
        ACT/30,,1/4,OUT;
        ACT/30,,3/4,EXT;
C5      QUEUE(31),0,1,BLOCK;
        ACT/31,,3/4,Q6;
        ACT/31,,1/4,EXT;
V6      QUEUE(32),0,1,BALK(PV6);
        ACT/32,,.75,Q6;
        ACT/32,,.25,V6;
B6      QUEUE(33),0,1,BALK(PB6);
        ACT/33,,.25,D6;
        ACT/33,,.75,B6;
I6      QUEUE(34),0,1,BALK(PI6);
        ACT/34,,.75,D6;
        ACT/34,,.25,I6;
D6      QUEUE(35),0,1,BLOCK;
        ACT/35,,.25,Q6;
        ACT/35,,.75,D6;
Q6      QUEUE(36),0,2,BLOCK;
        ACT/36,ATRIB(2);
        GOON,1;
        ACT,,ATRIB(1),EQ.6,O6;
        ACT,,ATRIB(1).GT.6,C6;
O6      QUEUE(37);
        ACT/37,,1/3,OUT;
        ACT/37,,2/3,EXT;
C6      QUEUE(38),0,1,BLOCK;
        ACT/38,,2/3,Q7;
```

```
         ACT/38,,1/3,EXT;
V7       QUEUE(39),0,1,BALK(PV7);
         ACT/39,,.75,Q7;
         ACT/39,,.25,I7;
B7       QUEUE(40),0,1,BALK(PB7);
         ACT/40,,.25,D7;
         ACT/40,,.75,B7;
I7       QUEUE(41),0,1,BALK(PI7);
         ACT/41,,.75,D7;
         ACT/41,,.25,I7;
D7       QUEUE(42),0,1,BLOCK;
         ACT/42,,.25,Q7;
         ACT/42,,.75,D7;
Q7       QUEUE(43),0,1,BLOCK;
         ACT/43,ATRIB(2);
         GOON,1;
         ACT,,ATRIB(1).EQ.7,O7;
* Only entities destined for node 7 are processed.
O7       GOON,1;
         ACT,,ATRIB(3).EQ.1,OI7;
* Only entities which originated at node 1 are processed.
OI7      QUEUE(44);
         ACT/44,,1/2,OUT;
         ACT/44,,1/2,EXT;
* 1/2 of entities processed are sent to the collection node and 1/2
are terminated.
         TERM;
OUT      COLCT,INT(4),NODE TIME;
* This node collects statistics on delay time and throughput for
processed entities which have an arrival code of 4.
         TERM;
PV1      COLCT,BETWEEN,PKT LOSS;
* Voice packet rejection for node 1 is collected here.
         TERM;
PB1      COLCT,BETWEEN,BPL1;
* Bulk packet rejection for node 1 is collected here.
         TERM;
PI1      COLCT,BETWEEN,IPL;
* Interactive packet rejection for node 1 is collected here.
         TERM;
PV2      COLCT,BETWEEN,VPL;
         TERM;
PB2      COLCT,BETWEEN,BPL;
         TERM;
PI2      COLCT,BETWEEN,IPL;
         TERM;
PV3      COLCT,BETWEEN,VPL;
         TERM;
PB3      COLCT,BETWEEN,BPL;
         TERM;
PI3      COLCT,BETWEEN,IPL;
         TERM;
PV4      COLCT,BETWEEN,VPL;
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        TERM;
PB4     COLCT,BETWEEN,BPL;
        TERM;
PI4     COLCT,BETWEEN,IPL;
        TERM;
PV5     COLCT,BETWEEN,VPL;
        TERM;
PB5     COLCT,BETWEEN,BPL;
        TERM;
PI5     COLCT,BETWEEN,IPL;
        TERM;
PV6     COLCT,BETWEEN,VPL;
        TERM;
PB6     COLCT,BETWEEN,BPL;
        TERM;
PI6     COLCT,BETWEEN,IPL;
        TERM;
PV7     COLCT,BETWEEN,VPL;
        TERM;
PB7     COLCT,BETWEEN,BPL;
        TERM;
PI7     COLCT,BETWEEN,IPL;
        TERM;
        END;
INIT,0,100;
* System run time for simulation model.
FIN;
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Voice and Data Integration, Telecommunications,
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This analysis determined the best approach and switching technique for use in DOD communications. The approach and switching techniques considered were integrated and nonintegrated voice and data networks using circuit, packet, and hybrid switching. The Simplified Voice Trunking Model(SVTM) from Rome Air Development Center was used as the network using hybrid switching for voice and data integration.

The analysis was accomplished with a network analysis of various performance criteria using different approaches to voice and data communication networks. The network analysis was performed using mathematical analysis and simulation models. The simulation models created were a circuit switched model for analog voice, a packet switched model for digital data, a packet switched model with multiple arrivals for digital voice, interactive data and bulk data, and a packet switched model for digital voice and interactive data. The simulation models allowed for changes in percentage of voice and data, arrival rates, service rates, path algorithm, percentage of interactive versus bulk data, queue lengths and distance of message travel. The results of the network analysis were also compared to the results of the Simplified Voice Trunking Model. The results of this analysis indicate that DOD needs to instigate intensive research into integrated voice and data communications using packet and hybrid switching techniques. This analysis showed the best approach to DOD communications was with use of voice and data integration and hybrid switching as used in the Simplified Voice Trunking Model.

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